RTTI

ROCKFORD TECHNICAL TRAINING INSTITUTE

1-800-

A PASSION FOR PERFORMANCE

Rockford Fosgate®

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Tempe, Arizona
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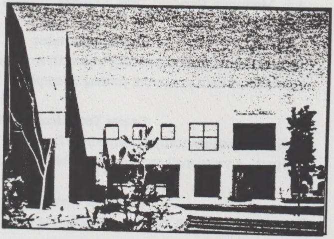
INTRODUCTION

Goals of Rockford Corporation

It is Rockford Corporation's mission to become the premier manufacturer of consumer electronics in the world.



To accomplish such an ambitious goal Rockford is aligning itself as a Market driven company. Literally translated; when the market changes Rockford has already anticipated the changes and aggressively develops new and innovative products to address current market trends. In addition, a "market driven" company responds to the needs of its dealers and consumers.

Rockford's success and growth will increase as we improve our ability to please the customer. Whether it's a store owner or a consumer with a problem, we strive to find solutions for the day to day challenges.

Rockford Fosgate.

WORLDWIDE REPRESENTATION

Rockford Services

Rockford Corporation **602-967-3565**

The Rockford Corporation offers professional dealer support to assist you with anything you may require. Rockford has three separate 800 telephone lines which provide immediate service from 8:00 a.m. to 5:00 p.m. Mountain Standard Time, Monday through Friday.

Sales Department **800-366-2349**

The Sales Department can assist you with any questions you may have with your account or any product information such as current pricing or employee purchase.

Dealer Technical Assistance **800-795-2382**

This line is available only to Authorized Rockford Fosgate Dealers and is maintained by Rockford's Technical Support. A voice mail service is available to leave a message during non-business hours.

Customer Service / Parts Dept. **800-669-9899**

This department can assist you in ordering parts and warranty repair on any Rockford Fosgate product.

Purpose for Sales Training Program

The primary purpose for this sales training program is to help you sell more Rockford Fosgate products. You'll learn the differences between our product and other brands, we know you will choose to sell Rockford Fosgate. You will also learn more about audio and become better skilled as a salesperson, increasing your income and the success of your business.

Many of the concepts we discuss are very basic. Since everyone in the audio business is at a different level of knowledge, the information presented is from its simplest form to the more advanced. This program is not designed to make engineers of salespeople. It is, however, designed to help you "Think on Your Feet." You'll be able to deal with objections and questions as if you've rehearsed each presentation prior to the customer walking in the door.

In order for you to get to this point, ***you must read this manual.*** Your sales trainer will highlight the most important points, but reading the entire manual will help you digest the information. Learn as much information as you can so that it can be used for your own benefit.

COMPANY HISTORY

Jim Fosgate Founder

Rockford Fosgate's beginning started with Jim Fosgate's dream of high performance car audio. When the first Fosgate PR-7000 automotive amplifier was shown at the Chicago C.E.S. in 1973, there weren't any head units on the market with line level outputs and there weren't any dealers willing to order a $300 car amp. This didn't stop Jim from pursuing his goals. He had found a better way and he was determined to educate the market.

Jim Fosgate's passion for audio was nurtured by a friend and avid hobbyist, a dealer who lived in the mid-west. While Fosgate's company Pro-Line in Salt Lake City was building radio transmitters and receivers for remote control airplanes, Jim Fosgate was developing a circuit called a "Frequency Energizer."

The "Frequency Energizer" circuit was developed for Fred Hulan's *Audio Mart* in Kansas City. Fred was doing professional installations in homes and theaters using a then state-of-the-art piece of test equipment called an "Acoust-A-Voice." It incorporated pink noise, a 1/3 octave equalizer and an oscilloscope. Fred kept notes on the inefficiencies he saw repeatedly in each installation. He shared those findings to Jim Fosgate whose Frequency Energizer circuit would compensate for loss in the high and low end frequencies. (Starting to sound familiar?) This first unit was designed to run through a tape loop in the receiver or preamplifier.

The birth of Fosgate car audio came when the Frequency Energizer was incorporated into a 30 watt car amp by Jim and some of his after hours enthusiasts. These late night experiments lead to the PR-7000.

Rockford Corporation

By the end of the seventies, Jim Fosgate had developed interests in other businesses and was pretty much letting the company run and manage itself. Financial troubles gave way to quality problems and Mr. Fosgate began looking for a buyer for his innovative car audio company.

Rockford Corporation was formed as an investment group. Their organization at the time was called Camelback Investment Group. They adopted the name of the street where Fosgate audio had existed for years in Tempe, Arizona. Their goal was to take a floundering company, turn it around, and make it an industry leader.

The biggest initial change made at the newly formed Rockford Fosgate was the strict reins put on quality control. Key people were put into positions and responsibilities that assured total quality control procedures. Many of the key individuals in these positions are still top executives at Rockford Corporation today.

One of these key people was an engineer named John France. John had actually been working with Jim Fosgate since the mid seventies. It wasn't until Rockford's involvement that John France's position became so important. It was John's interpretation of Jim Fosgate's designs that made Rockford Fosgate's early products so successful. Innovative ideas such as bridging mono while running stereo and the use of MOSFET's for better stability at low impedance loads are proof of John's vision. John also developed the protection circuit that is now used in all Punch and Series One amplifiers. This circuit was first developed for the Power 1000 and its effectiveness was demonstrated to R.T.T.I. students by burning pencils and welding with foil without damage to the amplifier.

As Rockford Fosgate products became more in demand, a peculiar problem developed. Dealers would call the factory and say, "We love your amps, but they're blowing up every speaker we have." So the next hurdle to overcome was to develop a high performance speaker line.

After evaluating many samples from outside speaker manufacturers, it became evident that nobody could deliver what Rockford Fosgate wanted. At that point it was determined that the only way to make a speaker worthy of the Rockford Fosgate name was to build it ourselves. Rockford found a quality O.E.M. speaker manufacturing in Grand Rapids, Michigan, which had been in the audio business since the early 1940's. Carbonneau speaker manufacturing had focused their business on the O.E.M. of mobile speakers. Carbonneau was excited about being involved with Rockford Corporation in the production of hi-performance car audio. In 1985, Rockford Fosgate began building its new line of car audio woofers, mid ranges, and tweeters. Rockford Carbonneau made a quick transformation from O.E.M. to after-market.

Rockford Technical Training Institute

In 1986 Rockford Corporation started a program to teach installers the best way to use Rockford Fosgate amplifiers and speakers. The course started with system design, passive crossover design and the basics of Speaker enclosures. A second, more advanced class was also developed and Rockford's Technical Training Institute was now in full scale operation. Still to this day there are those who can't believe that one amp can run twenty speakers. The Rockford Technical Training Institute helped Rockford dealers stay several steps ahead of the competition.

The R.T.T.I. program in its original form brought installers from around the country to Arizona for three and a half days of training. Not a bad deal if you came to a winter class and you lived in Wisconsin. This type of program began to get a little costly and kept students away from their stores too long. Today, Rockford Corporation uses a more effective program where some training is done in the field, and the more advanced training involving custom fabrication and installation continues to be taught at our R.T.T.I. facility in Tempe.

Attendance to either of the R.T.T.I. programs is restricted to Rockford Fosgate dealers only. The Road Show is scheduled a year in advance. Check with your factory representative for dates and locations. The advanced training in Tempe is reserved for installers chosen by the factory representative and regional sales manager.

Car Audio Contest Trilogy

Rockford Corporation has always believed that the more our industry prospers, the more Rockford will prosper. For this reason Rockford Corporation has always been a supporter of I.A.S.C.A. Rockford employees co-authored the first National Autosound Challenge Association (N.A.C.A.) judging sheet and rule book. This organization eventually became the International Autosound Challenge Association.

In 1985, Rockford Fosgate marketing wanted to take the concept of contests that was being promoted by Doc Thunder and put it into a format that would expand the awareness of quality high-end car audio.

With the help of John France, Rockford Fosgate created, outlined and defined the goals of sound quality, installation and weighted sound pressure level as determining factors of a winning system. This original sheet was very simple and easy to use. I.A.S.C.A. uses very much the same scoring sheet to this day.

Rockford Fosgate created the challenge package to help dealers learn how to promote contests. At this time Rockford Fosgate even supplied the judging sheets and the judges. Time went by and everyone saw the value of promoting through contests. Rockford Fosgate decided to share our experience in this sort of promoting to the entire industry in a meeting at the WCES. We felt it would benefit the entire country if all manufacturers supported the same judging format.

This led to the formation of N.A.C.A. and later I.A.S.C.A. We dreamed of our efforts turning into something like N.H.R.A., but to see it actually happen has been like watching history being written.

Team America Program (I.A.S.C.A / U.S.A.C.)

Rockford Fosgate announces an updated Team America Program designed to significantly improve support of competitors using Rockford Fosgate products. This program is open to anyone who purchases Rockford Fosgate equipment.

Team America members must be active members of IASCA or USAC. If the individual is not a member of one of these organizations, Rockford will provide applications to join. Team America members will receive a Team America t-shirt, product patch, Wearables brochure, car sticker, membership card, and the Team America Magazine, "Made in the U.S.A." Members can earn points towards several unique Team America products through competition in IASCA or USAC competition. If a Team America member qualifies and competes in either an IASCA or USAC National Final, that member will receive special prizes.

THE ROCKFORD EPIC

1969-1973	Jim Fosgate's company *Pro-Line* builds radio transmitters and receivers used in remote control airplane models. Audio is just a hobby.
1973	Jim Fosgate designs home EQ (Frequency Energizer) unit to compensate for inefficiencies in home and theater listening environment.
1973	Jim Fosgate's company, Pro-Line, puts Frequency Energizer into 30 watt per channel car amp. The patented Punch EQ circuit is born.
1973	Summer Consumer Electronics Show. The newly formed auto sound company, Fosgate Audio, showed a revolutionary amplifier with unique Punch EQ. First model was the PR 7000.
1975	Fosgate begins manufacturing amplifiers with discrete components rather than integrated circuits. John France is involved with engineering of new products.
1980	ROCKFORD CORPORATION is formed and invests in Jim Fosgate's company and ideas.
1980	Jim Fosgate leaves *Fosgate* to pursue other ventures including his radically innovative audio/video surround sound processing.
1982	Product quality & reputation improves dramatically.
1983	Thunder On Wheels holds first Rockford Fosgate *"Crank 'em Up"* in Texas.
1985	Rockford Fosgate introduces Punch separates and woofers.
1986	Rockford Technical Training Institute (R.T.T.I.) is formed.
1986	Rockford Corporation acquires Carbonneau Speakers, primarily an O.E.M. builder of speakers.
1986	Pro Series introduced at W.C.E.S.
1990	Rockford Fosgate introduces the Box That Rocks at W.C.E.S.
1991	Symmetry is unveiled at Winter Consumer Electronics Show.
1993	New Punch DSM amplifiers and RFA Audiophile midrange and tweeters unveiled at WCES.
1994	New Punch "i" and "ix" series amplifiers; SYMMETRY EPX; 3X / 5X Signal Processors; Power Series Components; Punch Classic Twins; Punch Classic Woofers and The New Box That Rocks introduced at WCES.

Chapter 1

System Design

SYSTEM DESIGN THE ROCKFORD WAY

(How Rockford Dealers can rise to the top)

SATISFY YOUR CUSTOMER'S NEEDS

To best satisfy the customer's expectations of his/her sound system, **LISTEN TO THE CUSTOMER**. Only by qualifying the customer's needs, can a dealer design a system that meets or exceeds the customer's expectations.

GOING THE EXTRA MILE

To separate Rockford dealers from the rest, maximize the performance of all systems regardless of system complexity to insure reliability and sonic performance.

BUILDING FOR THE FUTURE

To benefit the customer and to promote future sales, design systems with expandability in mind. Customer's money spent on equipment that cannot be used in future systems is money wasted.

BUILDING A SYSTEM

Listed below are some items to consider when designing a system for a customer.

☛ **SPEAKERS**

1. Size
2. Performance
3. Location and type of vehicle

☛ **AMPLIFIERS**

1. How much power is needed for the speakers
2. Current available from the battery
3. Compatibility for future system design

☛ **CROSSOVERS**

1. Passive or Electronic
2. 2-Way, 3-Way, or 4-Way System Design
3. Crossover frequencies
4. Subsonic filters

☛ **EQUALIZATION**

1. Passive or Electronic
2. 1/2 Octave or 1/3 Octave Electronic Equalization
3. Passive Networks

SYSTEM DESIGNS

In order to achieve a stereo image in an automobile some basic principles should be applied. Designing a system around the principle of front stage, rear fill will enable the listener to perceive a stereo image in front of them. The rear fill is the result of sound reflecting off the windshield and other surfaces which then travel to the listener.

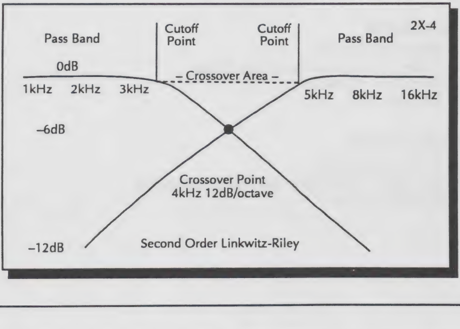
A front stage and stereo image can be achieved by proper speaker placement. Speakers need to be positioned on axis with the listener, which means speakers in doors may not always be the best location since you are relying on how well their off axis dispersion is.

We must also understand that one speaker alone cannot repoduce the human ear's audible bandwidth of 20Hz-20kHz. There is a critical bandwidth which is a three octave range centered around 1kHz. This bandwidth starts at 375Hz and ends at 3kHz and is where the human ear is most sensitive to amplitude, pitch, clarity and linearity.

Component speakers were introduced to car audio because of their efficiency in their limited bandwidth and mounting location flexibility. Each speaker has its own bandwidth where it plays naturally.

SUB BASS	MID BASS	MID RANGE	TREBLE
20Hz - 100Hz	100Hz - 275Hz	275Hz - 4kHz	4kHz - 20kHz

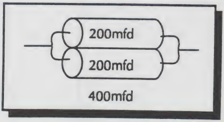
Component speakers must utilize a passive network in order to control their response and the resistance that the amplifiers sees. The total resistance of multiple speakers wired in parallel is relative to the crossover point of the speakers. Passive crossovers are not powered and are designed to react to a speaker's circuit. This reaction makes the network limit the frequencies to the speaker.



PARALLEL AND SERIES CAPACITORS AND COILS

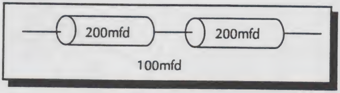
Capacitors

To calculate a certain frequency with two capacitors in parallel, add their total values. They should be non-polarized.



Capacitors in series create a lesser value. To calculate a certain frequency with two capacitors in series, use this formula:

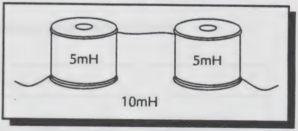
$$C_2 = \frac{1}{1/C_1 - 1/C_F}$$



C_1 is the value of the capacitor you have.
C_F is the capacitor value needed.
C_2 is the value required to get C_F.

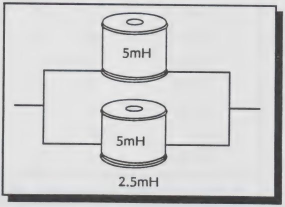
Coils

With coils in series, the values are added.



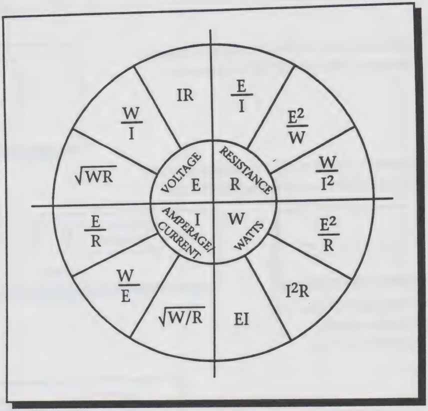
Coils in parallel create a lesser value. To calculate a certain frequency with two coils in series, use this formula:

$$L_2 = \frac{1}{1/L_1 - 1/L_F}$$



L_1 is the value of the coil you have.
L_F is the coil value needed.
L_2 is the value required to get L_F.

FORMULAS AND POWER TABLES



Power Cable Calculator								
Total Amperage Draw of System	**Up to 4 ft.**	**4 to 7 ft.**	**7 to 10 ft.**	**10 to 13 ft.**	**13 to 16 ft.**	**16 to 19 ft.**	**19 to 22 ft.**	**22 to 28 ft.**
0-20	14	12	12	10	10	8	8	8
20-35	12	10	8	8	6	6	6	4
35-50	10	8	8	6	4	4	4	4
50-65	8	8	6	4	4	4	4	2
65-85	6	6	4	4	2	2	2	0
85-105	6	6	4	2	2	2	2	0
105-125	4	4	4	2	0	0	0	0
125-150	2	2	2	0	0	0	0	0

The above chart shows wire gauges to be used, if no less than a .5 volt drop is accepted. If aluminum wire or tinned wire is used, the gauges should be of an even larger size to compensate.

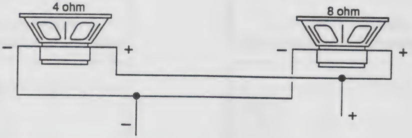
Cable gauge size calculation takes into account terminal connection resistance.

OHM'S LAW

Parallel Wiring Formula for 2 Speakers

$$\frac{R1 \times R2}{R1 + R2} = \text{LOAD IMPEDANCE}$$

EXAMPLE: $\dfrac{\text{4 Ohms times 8 Ohms}}{\text{4 Ohms plus 8 Ohms}} = \dfrac{32}{12} = 2.67$ Ohms



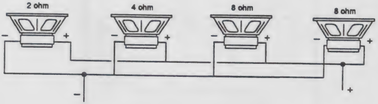
Parallel Wiring for Multiple Impedances

$$\frac{1}{\frac{1}{R1} + \frac{1}{R2} + \frac{1}{R3} + \frac{1}{R4}}$$

EXAMPLE:

Drivers' Impedance in Fraction form

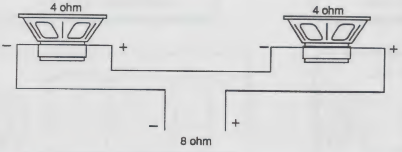
$$\frac{1}{\frac{1}{2} + \frac{1}{4} + \frac{1}{8} + \frac{1}{8}}$$



Convert to Decimal form and add together $\dfrac{1}{.5 + .25 + .125 + .125} = 1$ Ohm

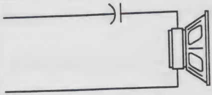
Series Wiring Formula

R1 + R2 = LOAD IMPEDANCE

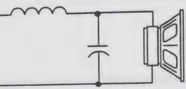


PASSIVE CROSSOVERS

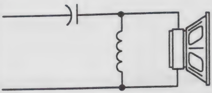
6dB/Octave Low Pass



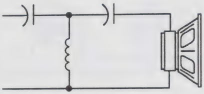
6dB/Octave High Pass



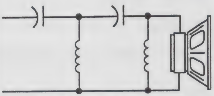
12dB/Octave Low Pass



12dB/Octave High Pass

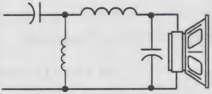


18dB/Octave High Pass

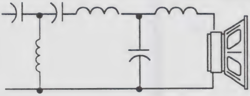


24dB/Octave High Pass

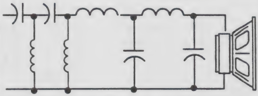
6dB/Octave Band Pass



12dB/Octave Band Pass

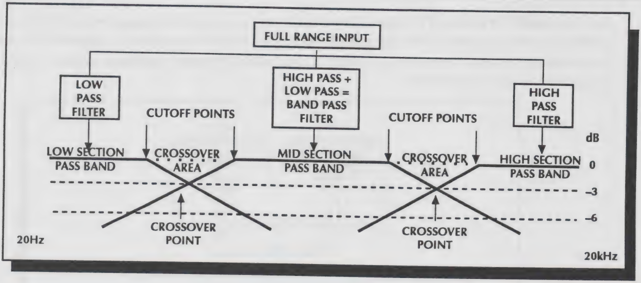


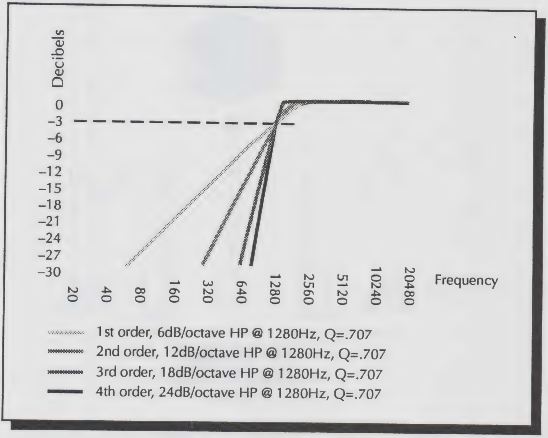
18dB/Octave Band Pass



24dB/Octave Band Pass

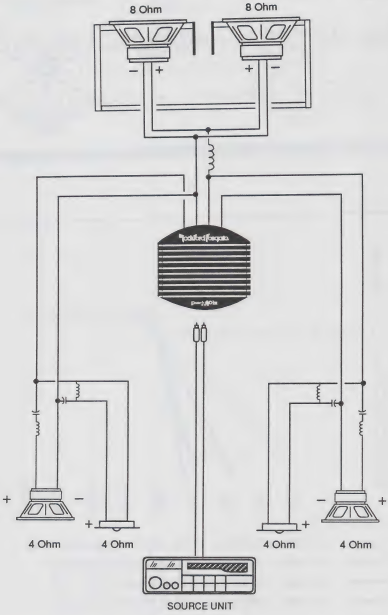
CROSSOVER SLOPES





3-WAY SYSTEM

The introduction to ***Rockford's System Design*** consists of one pair of midrange/tweeter and one pair of woofers. The midrange and tweeters are mounted close together and as far forward as possible. This develops our primary front stage. The woofers are always wired in mono to achieve maximum output from the amp. All speakers are wired in parallel.

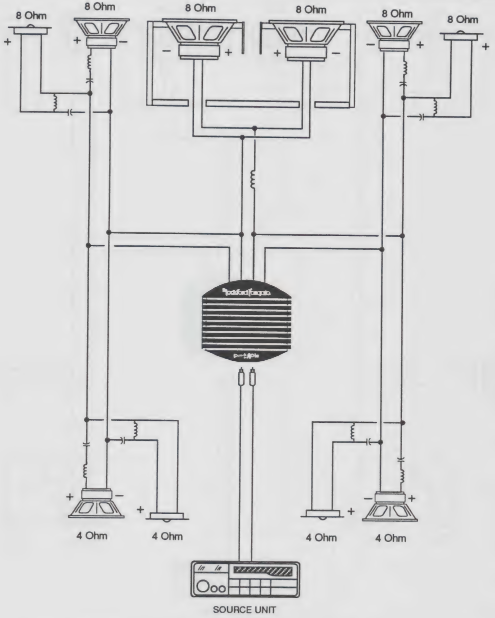


Total power applied to each bandwidth including all outputs:

Sub-bass	Midbass	Midrange	Treble	
120 Watts		30 Watts	30 Watts	RMS Power
212 Watts		67 Watts	67 Watts	PowerCube Dynamic Power at 12.6 Volts

3-WAY SYSTEM
FRONT STAGE REAR FILL

Rockford's 3-way system consists of 8 Ohm speakers mounted in the rear of the vehicle. 8 Ohm speakers will play 3dB quieter than 4 Ohm and will simulate rear ambiance. All speakers are wired in parallel, lowering the overall impedance.

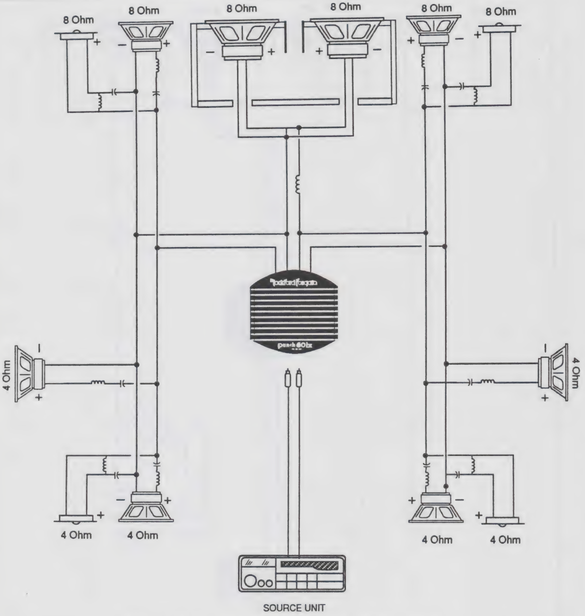


Total power applied to each bandwidth including all outputs:

Sub-bass	Midbass	Midrange	Treble	
120 Watts		60 Watts	60 Watts	RMS Power
212 Watts		102 Watts	102 Watts	PowerCube Dynamic Power at 12.6 Volts

4-WAY SYSTEM
FRONT STAGE REAR FILL

To reinforce a stronger front stage, a midbass driver will be required. The mounting location must be in front of the car and requires a bandpass crossover. In this 4-way system the listener will perceive that bass is coming from in front of the car. All speakers are wired in parallel.

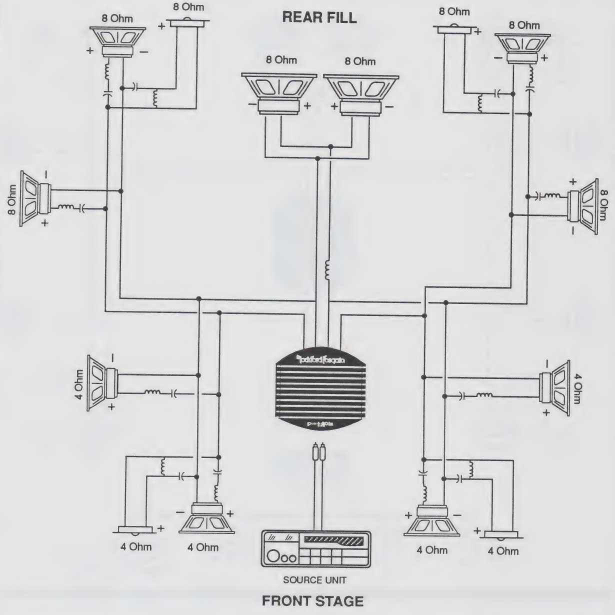


Total power applied to each bandwidth including all outputs:

Sub-bass	Midbass	Midrange	Treble	
120 Watts	24 Watts	48 Watts	48 Watts	RMS Power
212 Watts	40 Watts	81 Watts	81 Watts	PowerCube Dynamic Power at 12.6 Volts

ULTIMATE 4-WAY SYSTEM

This ultimate Rockford system makes an impressive demonstration vehicle to show customers the flexibility of Rockford Fosgate amplifiers. The design consists of 14 speakers running off of one single amplifier. The woofers are wired in parallel and each woofer is 8 Ohms. Also added to this system is a pair of 8 Ohm midbass for rear fill.

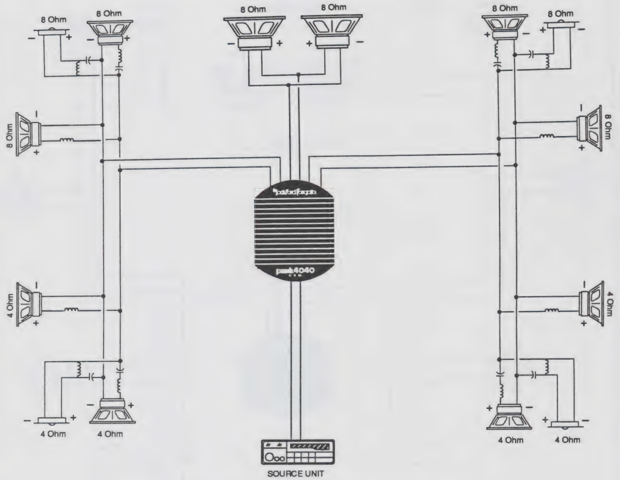


Total power applied to each bandwidth including all outputs:

Sub-bass	Midbass	Midrange	Treble	
120 Watts	40 Watts	40 Watts	40 Watts	RMS Power
212 Watts	68 Watts	68 Watts	68 Watts	PowerCube Dynamic Power at 12.6 Volts

ULTIMATE 4-WAY SYSTEM

Rockford's new 4 channel amplifier has a lot of flexibility with internal crossovers and separate equalization. This system is designed with passive components using electronic crossover assist on subwoofers and high pass on components at 100Hz, 12dB per octave. The 4040DSM is in 3-channel mode.

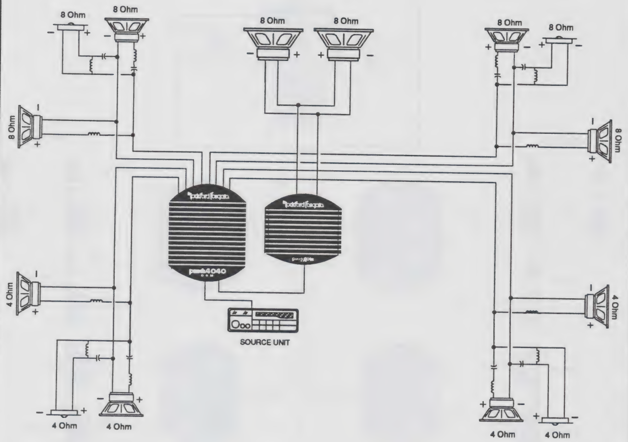


Total power applied to each bandwidth including all outputs:

Sub-bass	Midbass	Midrange	Treble	
80 Watts	26 Watts	26 Watts	26 Watts	RMS Power
90 Watts	29 Watts	29 Watts	29 Watts	PowerCube Dynamic Power at 12.6 Volts

ULTIMATE 4-WAY SYSTEM

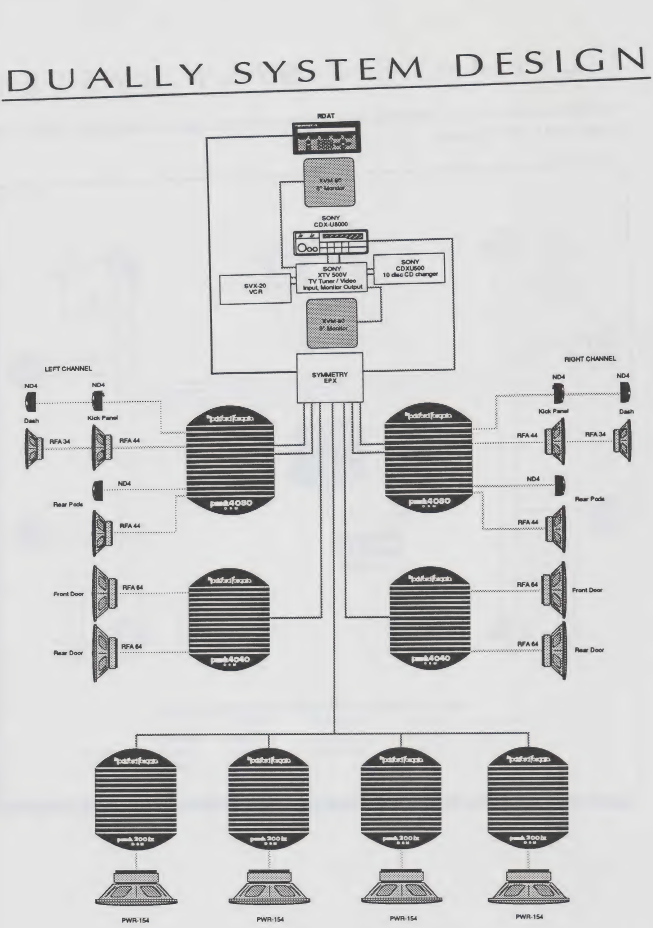
The 4040DSM is in 4-channel mode with 100Hz high-pass and has a signal out to feed into the Punch 60ix bridged with a 100Hz low-pass.



Total power applied to each bandwidth including all outputs:

Sub-bass	Midbass	Midrange	Treble	
120 Watts	26 Watts	26 Watts	26 Watts	RMS Power
212 Watts	34 Watts	34 Watts	34 Watts	PowerCube Dynamic Power at 12.6 Volts

DUALLY SYSTEM DESIGN

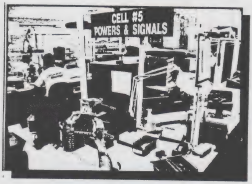


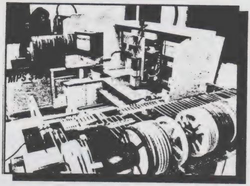
Chapter 2

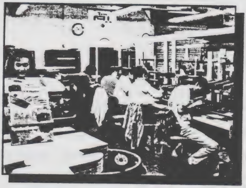
Amplifiers

Computerized Burn-In & Test Procedures

- Automated Component Insertion
- All Specifications Checked for Slightest Deviation
- Burn-In Stress Conditions to Find Defective Parts
- Rubber Mallet Tested For Loose Components or Cold Solder Joints





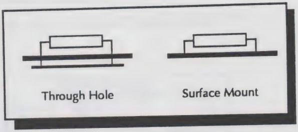


Installation Features

- Simultaneous Mono/Stereo Operation
- Variable Low to High Level Input Controls
- Gold Plated RCA Inputs
- Efficient Heat Sink for Proper Cooling

DSM Discrete Surface Mount

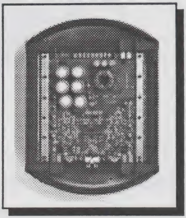
Amplifier manufacturing process that utilizes a state-of-the-art pick and place machine that checks tolerance and value of small discrete components which maximize performance while minimizing PC board space usually required.



Feature Discrete Components
Function One Component, One Task
Benefit Better control over task

Feature Surface Mount
Function Install components on surface of PC board as opposed to through-hole components
Benefit By using the pick and place machine, the automated manufacturing process reduces connections and crosstalk, lowers operating temperature, and increases reliability by utilizing superior components such as: metal film resistors, due to increased demand of superior components by other industries (aerospace, automotive, and computer).

Feature Large Trace Widths
Function Conducts more current
Benefit Greater output, power at lower speaker impedances



Feature Metal Film Resistors
Function Assist op-amps which amplify input signal
Benefit Lower tolerance

Adjustable Electronic Crossover ("X")

A more efficient way to process the signal before it reaches the amplifier stage.

Feature Electronic Crossover

Function To filter the input signal before amplification

Benefit More control with less distortion and higher efficient amplifier operation

MOSFET Modulated Power Supplies

These devices supplies the current needed to operate the active devices that drive the amplifier stage from the rectification process of the power supplies.

Feature MOSFET Power Supply

Function Produce higher voltage while drawing less current

Benefit MOSFET transistors are used with lower heat and more output because of their inherent efficiency to transfer larger amount of current that bipolar can deliver

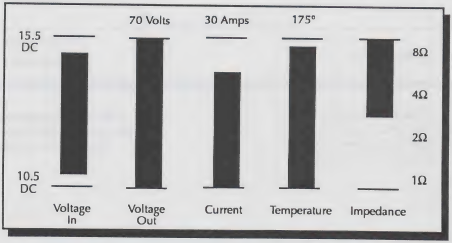
RTP Real Time Protection

An analog computer that senses temperature. The instantaneous current/voltage in the output stage so there is no need for internal fusing.

Feature Real Time Protection

Function To monitor voltage, current, temperature

Benefit Process happens now instead of later, no time shifting because An analog computer protects the amplifier at predetermined threshholds unlike current limiting which could cause premature protection and without the use of internal fusing.



Independent Left/Right Gain Control

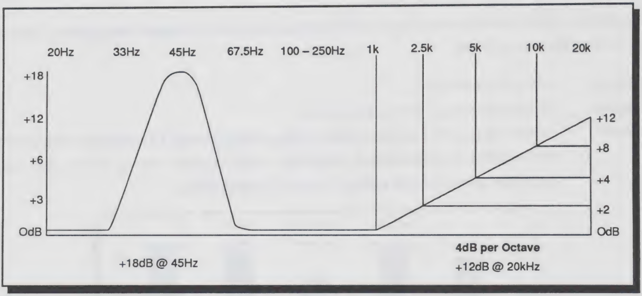
A more accurate level matching between source unit outputs to the inputs of the amplifier.

Feature	Independent Gain Control
Function	Independent level adjustment from the source to two channels
Benefit	Equal level in output between two channels

Punch Equalization

Compensate for the inadequacies of a vehicle's acoustical response.

Feature	Punch Input Module
Function	18dB boost at 45Hz
	12dB boost at 20kHz
Benefit	Increased bass and high frequency response utilizing peak and shelving type equalization

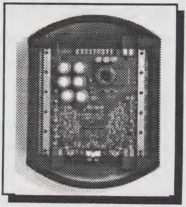


Peak Boost Equalization
with increased Q relative to level.
Qs equal to the center frequency divided by the bandwidth.

Shelving Type Equalization
Boost increase per octave from the fundamental frequency.

"i" Series

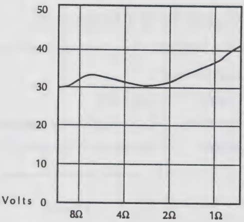
The use of surface mount technology enables 98% discrete components to be mounted directly to the PC board so the need of modules are eliminated.



Feature	"i" series
Function	Incorporating 98% discrete components that mount directly to the PC board
Benefit	Less connections for a shorter signal path and increased signal to noise ratio

CRV Controlled Rail Voltage

Automatic regulation of the PWM circuit maintains control over the rail voltage enabling a lower output impedance and increase of power output at lower impedances.

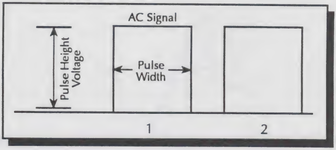


Feature	Controlled Rail Voltage
Function	To optimize the rail voltage at any given spekaer impedance
Benefit	No current limiting at lower impedances for increased low bass impact.

Pulse Width Modulation

This circuit varies the width of the DC/DC converter drive pulse to control the second AC signal and the amount of voltage produced by the power supply.

Feature	PWM
Function	A signal encoding process that regulates the power supply voltage relative to musical content and the vehicle's voltage fluctuation
Benefit	Increased dynamic headroom with continuous signal level



N-Channel MOSFET Outputs

Matched transistor devices that have an excessive amount of electrons from a doner material applied to the collector. These transistor conduct the power capability of the amplifier.

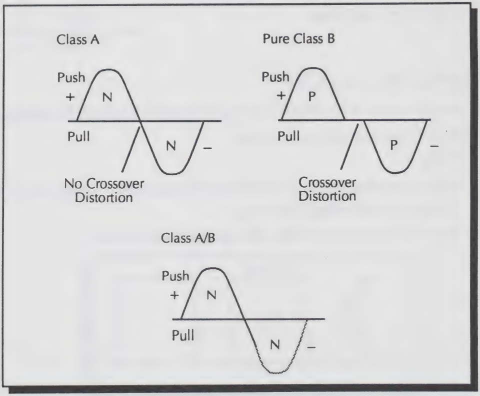
Feature	N-Channel
Function	Balanced push-pull switching
Benefit	Matched components for increased efficiency and no crossover distortion

Feature	MOSFET Outputs
Function	To deliver current to various impedance loads
Benefit	Stable into low impedances with low distortion and high thermal stability

Class A/B Operation

Two transistors in a push-pull configuration. One amplifying positive wave the other amplifying the negative wave.

Feature	Class A/B
Function	Minimum biased transistors so neither transistor turns completely off
Benefit	No crossover distortion, increased linearity



Cast Aluminum Heatsink

This type of extrusion has 30% more thermal time over an aluminum extrusion.

Feature	Cast Aluminum
Function	Transmits heat more effectively
Benefit	A lower internal operating temperature
Feature	Heatsink
Function	To absorb heat generated from internal circuitry
Benefit	Increased reliability and efficiency with a unique heatsink design

Interlocking End Caps

Designed to give a clean cosmetic appeal by covering wiring and mounting screws.

Feature	Interlocking
Function	Molded connection to heatsink via allen screws
Benefit	Not easily accessible for consumer alterations to control
Feature	End Caps
Function	To cover controls, wiring and mounting screws
Benefit	Highlighted mounting and protection against theft

THE AUDIOGRAPH / POWERCUBE

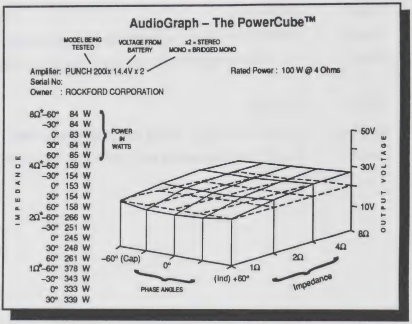
"Why Do We Test This Way?"

Amplifier testing has been used to measure the capabilities of amplifiers. The institute of high fidelity (IHF 202) test is recognized by manufacturers and is considered the industry standard in power ratings and has been used in car audio for numerous years.

The IHF-202 test consists of a 1kHz bursted tone which is pulsed on for 20 ms and then attenuated 20dB for 480 ms at a constant voltage (14.4 volts).

Some manufacturers give high power ratings but also have a high distortion rating sometimes 5% or higher total harmonic distortion. The power ratings given are from a purely resistive load. This is not the way an amplifier deals with music. Music is dynamic and is constantly changing.

The AudioGraph/PowerCube is a test instrument from Sweden that Rockford has utilized to establish power ratings that illustrate how an amplifier reacts to complex signals and inductive and capacitive loads of a speaker. The PowerCube is used to show that all amplifiers are not created equal and a Rockford amplifier will meet or exceed its rating from 8 Ohms to 1 Ohm. The PowerCube follows the same bursted tone at 1kHz with 20 ms on and 480 ms attenuated to 20dB. The 1kHz sinewave has .1% THD and the level is variable from 0 volts to 4 volts with 1 millivolt steps.



The maximum dynamic power is plotted on the graph and has a distortion of 1%. If any output exceeds 1% in distortion, a zero will appear on the graph to show that it is not satisfactory. Rockford tests their amplifiers in stereo and in mono at 14.4 volts and 12.6 volts. This can illustrate an ideal voltage and the static voltage of a vehicle's battery with the engine not running.

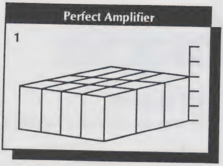


Figure 1. Perfect Amplifier
This amplifier works as a perfect voltage generator and is not affected by the connected load.

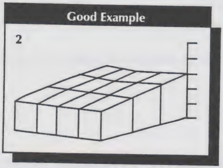


Figure 2. Good Example
This is an amplifier with typically good behavior. There are some low impedances losses, but the PowerCube is still almost cubic.

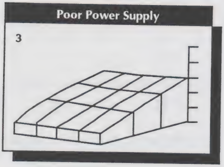


Figure 3. Poor Power Supply
This amplifier's power supply is too small or has too few output devices and cannot produce enough current in low impedances. As a result, the dynamic output power gets lower in these impedances.

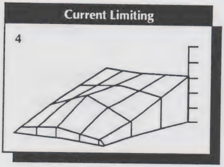


Figure 4. Current Limiting
This amplifier probably has too few output devices. To protect them, the amplifier has an electronic protection circuit which turns off the amplifier at "dangerous loads."

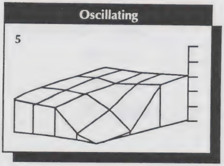


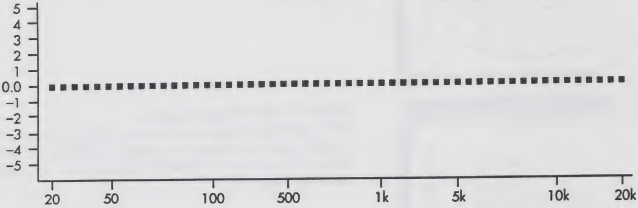
Figure 5. Oscillating
This amplifier is not stable at certain reactive loads. As a result, the distortion is always over 1% THD and the resulting output power is 0 watts.

Rockford Fosgate
Punch Verification Certificate

Date: 01-05-1994
Model: Punch 200ix
Serial #: B1THA4A0000003

Test Impedance: 2 Ohms
Test Voltage: 14.2 Volts DC

ACTUAL FREQUENCY RESPONSE PLOT



ACTUAL TEST MEASUREMENTS

TEST SEQUENCE	TEST PARAMETERS MIN	MAX	ACTUAL	
Idle Current	0.15	5.0	1.22	Amps Idle
Loaded Current	35.0	65.0	55.0	Amps Draw
Turn-on Delay	1.0	3.0	1.0	Seconds
Efficiency	40.0	—	76.9	Percentage
Output Voltage	18.0	—	24.5	Volts RMS
Output Power	162.0	—	300.5	Watts/Channel
Tracking Error	—	0.8	0.2	dB
Channel Separation		—	>100dB	dB
Signal to Noise		—	>100dB	dB

TEST RESULTS PASSED

Punch Verification Certificate
Explanation of Terms

Date: The date the amplifier was built.

Model / Serial Number: Every amplifier is tested. This assures Rockford Fosgate 100% quality control on the PC board.

Test Impedance / Test Voltage: A purely resistive 2Ω load.
The *Test Voltage* is representative to what a properly working automotive charging system will provide the amplifier allowing it to make its rated *Output Power.*
Watts per Channel = Volts²/Impedance.

Actual Frequency Response Plot: This test shows the exact frequency response of each amplifier.

Actual Test Measurements
Minimum/Maximum/Actual:

Idle Current: The amount of current the amplifier is drawing from the car with no load applied.

Loaded Current: This test shows how much current the amplifier will draw with a load applied. This is relative to impedance and voltage. With the same voltage in a 4Ω application this amplifier will draw half the current than it will at 2Ω. This test is very important when trying to determine the total amperage draw of a multi-amp system.

Turn-On Delay: How many seconds before the amplifier un-mutes itself. This circuit eliminates turn-on and turn-off pops.

Efficiency: Efficiency is the ability of the amplifier to convert DC input voltage to useable AC output voltage.

Output Voltage / Output Power: Output voltage and impedance (2Ω in this test) are used to calculate Output Power (watts per channel) RMS power is what the amplifier will do continuously, even under the worst conditions. Dynamic power is what the amplifier has on "Reserve." Rockford amplifiers will DOUBLE their rated output (even in MONO) for short periods. This gives you +3dB of Dynamic head room.

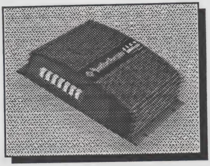
Tracking Error: Each channel is tested separately to track the level between one channel and the other resulting in a dB measurement.

Channel Separation: A measurement of the isolation of one channel from another on a dB scale.

Signal to Noise: A measurement weighted or non-weighted to show the ratio of audio signal relative to the nosie floor.

SERIES 1 AMPLIFIERS

FEATURES

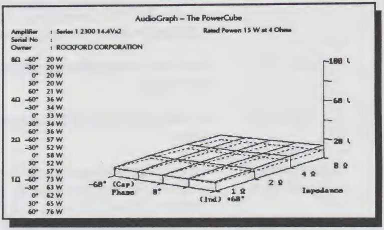


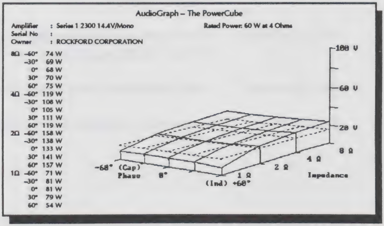
2300

- ☞ 15 Watts x 2 into 4Ω .08% THD
- ☞ 30 Watts x 2 into 2Ω .30% THD
- ☞ 60 Watts bridged into 4Ω .30% THD

EXCLUSIVE ROCKFORD FEATURE

- 3 YEAR WARRANTY
- "i" SERIES
- DSM DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- ☞ N-CHANNEL / OUTPUT CIRCUITRY
- "NEW" ☞ TERMINAL CONNECTORS
- "NEW" ☞ HEATSINK DESIGN
- ☞ HIGH LEVEL INPUT
- ☞ GOLD-PLATED CONNECTORS

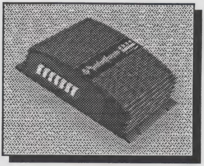




SERIES 1 AMPLIFIERS

FEATURES

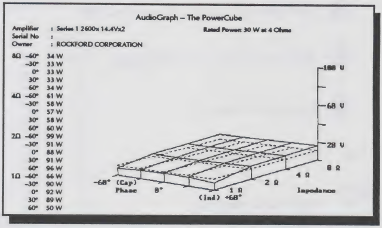
2600X

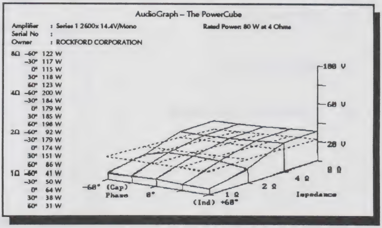


- 30 Watts x 2 into 4Ω .08% THD
- 60 Watts x 2 into 2Ω .30% THD
- 120 Watts bridged into 4Ω .30% THD

EXCLUSIVE ROCKFORD FEATURE

- 3 YEAR WARRANTY
- "i" SERIES
- DSM DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- ELECTRONIC CROSSOVER
- N-CHANNEL / OUTPUT CIRCUITRY
- "NEW" TERMINAL CONNECTORS
- "NEW" HEATSINK DESIGN
- HIGH LEVEL INPUT
- GOLD-PLATED CONNECTORS

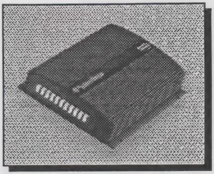




SERIES 1 AMPLIFIERS

FEATURES

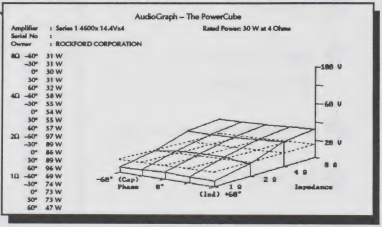
4600X

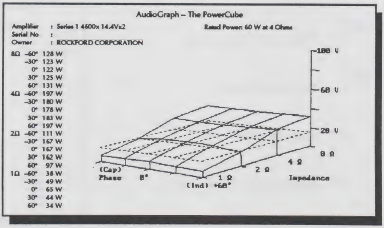


- 30 Watts x 2 into 4Ω .08% THD
- 60 Watts x 2 into 2Ω .30% THD
- 120 Watts bridged into 4Ω .30% THD

EXCLUSIVE ROCKFORD FEATURE

- 3 Year Warranty
- "i" Series
- DSM Discrete Surface Mount Compon
- RTP Real Time Protection Circuit
- Electronic Crossover
- N-Channel / Output Circuitry
- "NEW" Terminal Connectors
- "NEW" Heatsink Design
- High Level Input
- Gold-Plated Connectors

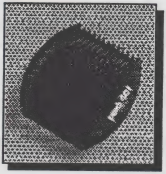




PUNCH AMPLIFIERS

FEATURES

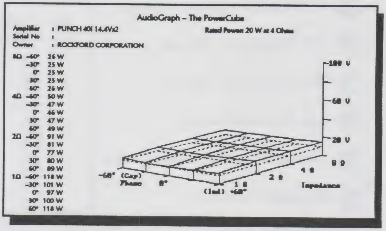
PUNCH 40i DSM



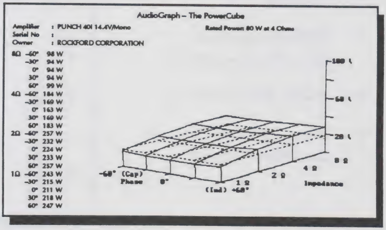
- 20 Watts x 2 into 4Ω .05% THD
- 40 Watts x 2 into 2Ω .1% THD
- 80 Watts mono into 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

- 3 YEAR WARRANTY
- "i" SERIES
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- CRV CONTROL RAIL VOLTAGE
- PIM PUNCH INPUT MODULE
- CAST ALUMINUM HEATSINK
- INTERLOCKING CAST ALUMINUM ENDCAPS

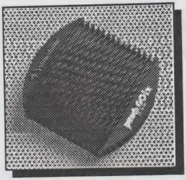


- N-CHANNEL V-FET
- MOSFET PULSE WIDTH MODULATED POWER SUPPLIES
- TERMINAL BLOCK CONNECTORS
- DOUBLE SIDED PLATED THROUGH GLASS EPOXY PC BOARD



PUNCH AMPLIFIERS

FEATURES



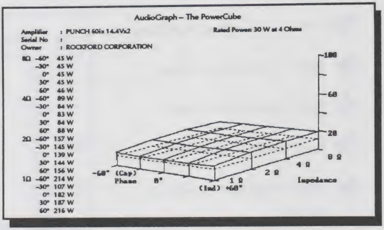
PUNCH 60ix DSM

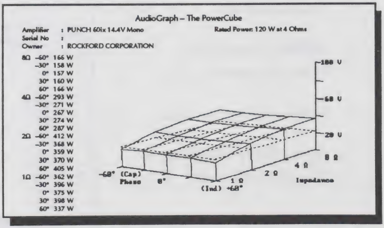
- 30 Watts x 2 into 4Ω .05% THD
- 60 Watts x 2 into 2Ω .1% THD
- 120 Watts mono into 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

- 3 YEAR WARRANTY
- "i" SERIES
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- CRV CONTROL RAIL VOLTAGE
- PIM PUNCH INPUT MODULE
- CAST ALUMINUM HEATSINK
- INTERLOCKING CAST ALUMINUM ENDCAPS
- "NEW" ELECTRONIC CROSSOVER

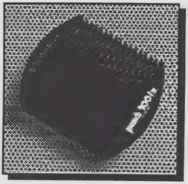
- N-CHANNEL V-FET
- MOSFET PULSE WIDTH MODULATED POWER SUPPLIES
- TERMINAL BLOCK CONNECTORS
- DOUBLE SIDED PLATED THROUGH GLASS EPOXY PC BOARD





PUNCH AMPLIFIERS

FEATURES

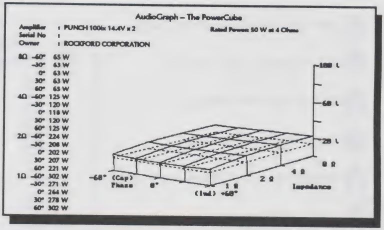


PUNCH 100ix DSM

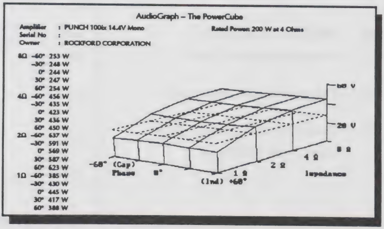
- 50 Watts x 2 into 4Ω .05% THD
- 100 Watts x 2 into 2Ω .1% THD
- 200 Watts mono into 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

- 3 YEAR WARRANTY
- "i" SERIES
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- CRV CONTROL RAIL VOLTAGE
- PIM PUNCH INPUT MODULE
- CAST ALUMINUM HEATSINK
- INTERLOCKING CAST ALUMINUM ENDCAPS
- "NEW" ELECTRONIC CROSSOVER



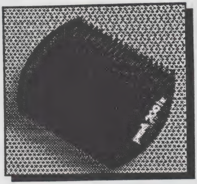
- N-CHANNEL V-FET
- MOSFET PULSE WIDTH MODULATED POWER SUPPLIES
- TERMINAL BLOCK CONNECTORS
- DOUBLE SIDED PLATED THROUGH GLASS EPOXY PC BOARD



PUNCH AMPLIFIERS

FEATURES

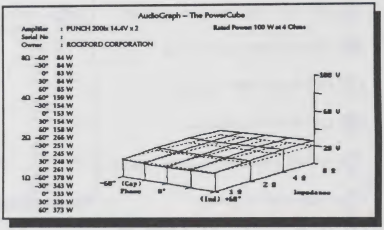
PUNCH 200ix DSM

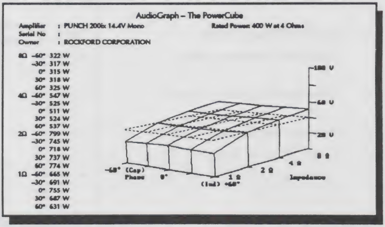


- 100 Watts x 2 into 4Ω .05% THD
- 200 Watts x 2 into 2Ω .1% THD
- 400 Watts mono into 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

- 3 YEAR WARRANTY
- "i" SERIES
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- CRV CONTROL RAIL VOLTAGE
- PIM PUNCH INPUT MODULE
- CAST ALUMINUM HEATSINK
- INTERLOCKING CAST ALUMINUM ENDCAPS
- "NEW" ELECTRONIC CROSSOVER
- N-CHANNEL V-FET
- MOSFET PULSE WIDTH MODULATED POWER SUPPLIES
- TERMINAL BLOCK CONNECTORS
- DOUBLE SIDED PLATED THROUGH GLASS EPOXY PC BOARD

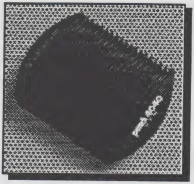




PUNCH AMPLIFIERS

FEATURES

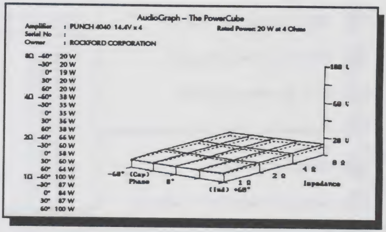
PUNCH 4040 DSM

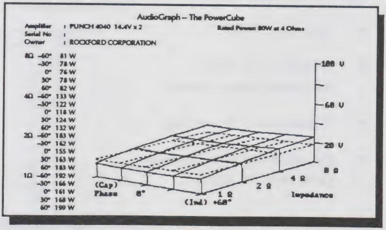


- 20 Watts x 4 into 4Ω .05% THD
- 40 Watts x 4 into 2Ω .1% THD
- 80 Watts x 2 into 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

- 3 YEAR WARRANTY
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- CRV CONTROL RAIL VOLTAGE
- PIM PUNCH INPUT MODULE
- CAST ALUMINUM HEATSINK
- INTERLOCKING CAST ALUMINUM ENDCAPS
- "NEW" ELECTRONIC CROSSOVER
- "NEW" SEPARATE PREAMP OUTPUT W/XOVER
- "NEW" POLARITY SWITCH
- N-CHANNEL V-FET
- MOSFET PULSE WIDTH MODULATED POWER SUPPLIES
- TERMINAL BLOCK CONNECTORS
- DOUBLE SIDED PLATED THROUGH GLASS EPOXY PC BOARD

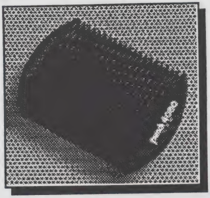




PUNCH AMPLIFIERS

FEATURES

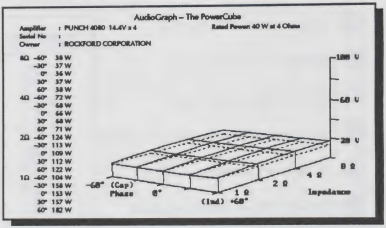
PUNCH 4080 DSM

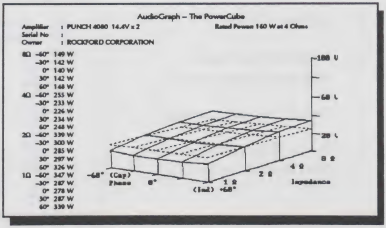


- 40 Watts x 4 into 4Ω .05% THD
- 80 Watts x 4 into 2Ω .1% THD
- 160 Watts x 2 into 4Ω .1% THD

EXCLUSIVE ROCKFORD FEATURE

- 3 YEAR WARRANTY
- DSM 98% DISCRETE SURFACE MOUNT COMPONENT
- RTP REAL TIME PROTECTION CIRCUIT
- CRV CONTROL RAIL VOLTAGE
- PIM PUNCH INPUT MODULE
- CAST ALUMINUM HEATSINK
- INTERLOCKING CAST ALUMINUM ENDCAPS
- "NEW" ELECTRONIC CROSSOVER
- "NEW" SEPARATE PREAMP OUTPUT W/XOVER
- "NEW" POLARITY SWITCH
- N-CHANNEL V-FET
- MOSFET PULSE WIDTH MODULATED POWER SUPPLIES
- TERMINAL BLOCK CONNECTORS
- DOUBLE SIDED PLATED THROUGH GLASS EPOXY PC BOARD





Chapter 3

Speakers

Kevlar™ Reinforced Spruce Pulp Cones

A speaker coating that strengthens the cones.

Feature	Kevlar
Function	Strengthen the cone for a linear excursion
Benefit	Less cone distortion and strong, low frequency response

Woven Kevlar™ Cone

This material is used in cone assemblies to give strength for increased power handling

Feature	Composite Kevlar
Function	To reinforce the once for a greater linear output
Benefit	Less cone distortion impervious to U.V. and temperature fluctuation

Dual Laminated Foam Surround

Assists in the suspension of the cone for a more linear cone movement.

Feature	Cone Movement
Function	Dual Laminated Foam Surround
Benefit	The dual laminated foam surround will control the woofer's excursion with lower surround resonances.

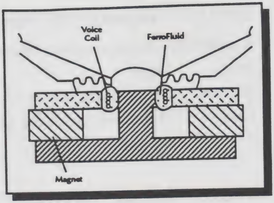
Nitrile Rubber Surround

Offers consistent control of the cone with more tolerance to heat and cold.

Feature	Nitrile Rubber Surround
Function	To control the movement of the cone and lower edge resonances
Benefit	Flat response and increased excursion

FerroFluid™

A magnetic fluid that is injected into the voice coil gap.

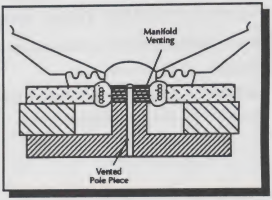


Feature	FerroFluid
Function	To cool the voice coil
Benefit	Increased efficiency and a lower impedance rise from less heat with a smooth frequency response

Proprietary Manifold Venting™

Eliminates pressure build-up behind the ferrofluid eliminating any loss.

Feature	Manifold Venting™
Function	Relieve pressure above and below the gap by the use of cross venting
Benefit	No loss of ferrofluid



Vented Pole Piece

Air ventilation through the former for cooling of the voice coil.

Feature	Vented Pole Piece
Function	To assist in cooling the former and voice coil
Benefit	Better power handling with less resistance introduced from heat

Neodymium Magnet

This metal is highly ionic and can hold a higher charge over cobalt ferrite magnet.

Feature	Neodymium Magnet
Function	To supply magnetic field in order to excite the movement of the speaker
Benefit	Smaller in size and highly efficient as opposed to ferrite magnets

PVA Treated Cones

The PolyVinyl Emulsion Treatment on cones increase performance with less weight to the cone and extended frequency response

Feature PVA (PolyVinyl Emulsion)

Function Strengthen and protect the cone with less cone flex

Benefit Smooth midrange and deep bass

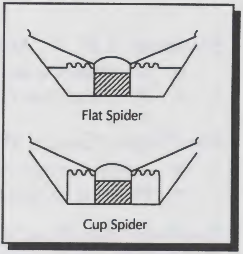
Flat Spider

The flat spider is designed to ensure proper cone control at the former.

Feature Flat Spider

Function To keep voice coil centered in the magnetic gap

Benefit Increased linearity and power handling



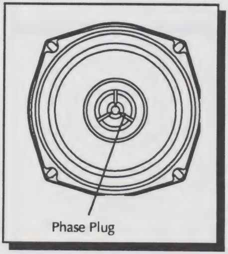
Phase Plug

Helps dispersion and increased linear off axis response

Feature Phase Plug

Function Control dispersion of tweeter

Benefit Extended high frequencies



SERIES 1 SPEAKERS

FEATURES

FULL RANGE OPERATING CAPABILITY

DIN SIZED BASKETS FIT A WIDE RANGE OF IMPORT VEHICLES 5.25" AND 6" MODELS

STANDARD 3.5", 4", 4x6" AND 6x9" BASKETS TO FIT DOMESTIC CARS

- 6"x9" Two Way and Three Way
- 3", 4", 4"x6" Dual Cone Design

LOW PROFILE GRILLES WITH LOGO SCREENED ON RING

- Packaged in Pairs with Grilles (excluding 3.5" and 4"x6")
- Packaging Includes Mounting Hardware and Speaker Wires

ONE PIECE BASE PLATE AND POLE PIECE ASSEMBLY

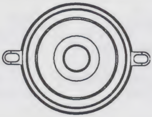
- Increases Magnetic Flux in Voice Coil Gap
- Increases Efficiency

DESIGNED FOR INFINITE BAFFLE MOUNTING

SERIES 1 SPEAKERS

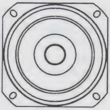
SPECIFICATIONS

S1-32



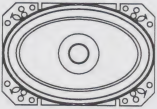
Diameter:	3.5"
Frequency Range:	150Hz-20kHz
Impedance:	4Ω
Sensitivity:	86.5dB w/m
Power Handling:	30 / 60 Watts
Series:	Two Way

S1-42



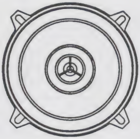
Diameter:	4"
Frequency Range:	115Hz-20kHz
Impedance:	4Ω
Sensitivity:	90dB w/m
Power Handling:	30 / 60 Watts
Series:	Two Way

S1-462



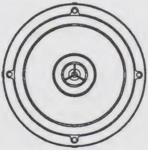
Diameter:	4"x6"
Frequency Range:	115Hz-20kHz
Impedance:	4Ω
Sensitivity:	90dB w/m
Power Handling:	30 / 60 Watts
Series:	Two Way

S1-5



Diameter:	5.25"
Frequency Range:	90Hz-20kHz
Impedance:	4Ω
Sensitivity:	89dB w/m
Power Handling:	50 / 100 Watts
Series:	Two Way

S1-6

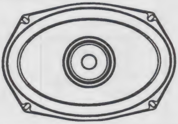


Diameter:	6"
Frequency Range:	85Hz-20kHz
Impedance:	4Ω
Sensitivity:	91dB w/m
Power Handling:	50 / 100 Watts
Series:	Two Way

SERIES 1 SPEAKERS

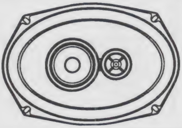
SPECIFICATIONS

S1-692



Diameter:	6"x9"
Frequency Range:	85Hz-20kHz
Impedance:	4Ω
Sensitivity:	93dB w/m
Power Handling:	50 / 100 Watts
Series:	Two Way

S1-693



Diameter:	6"x9"
Frequency Range:	60Hz-20kHz
Impedance:	4Ω
Sensitivity:	93dB w/m
Power Handling:	50 / 100 Watts
Series:	Two Way

PUNCH CLASSIC TWINS

FEATURES

KEVLAR REINFORCED SPRUCE PULP CONES

DUAL LAMINATED FOAM SURROUND

NEODYMIUM MAGNET (ON TWEETER)

(D) EURO DIN SNAP-IN VERSION

LARGER VOICE COIL

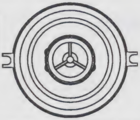
- ☞ 1" (34T2, 44T2)
- ☞ 1.25" (64T2)
- ☞ 1.5" (54T2, 694T2)

PHASE PLUG

PUNCH CLASSIC TWINS

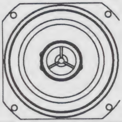
SPECIFICATIONS

PCH-34T2/ PCH-34T2D



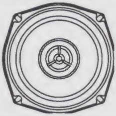
Diameter:	3.5"
Frequency Range:	175-20,000 Hz
Impedance:	4Ω
Sensitivity:	85dB w/m
Power Handling:	50 / 100 Watts max.
Series:	Two Way

PCH-44T2 PCH-44T2D



Diameter:	4"
Frequency Range:	100-20,000 Hz
Impedance:	4Ω
Sensitivity:	90dB w/m
Power Handling:	50 / 100 Watts max.
Series:	Two Way

PCH-54T2 PCH-54T2D

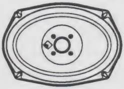


Diameter:	5.25"
Frequency Range:	85-20,000 Hz
Impedance:	4Ω
Sensitivity:	90dB w/m
Power Handling:	80 / 160 Watts max.
Series:	Two Way

PCH-64T2

Diameter:	6"
Frequency Range:	80-20,000 Hz
Impedance:	4Ω
Sensitivity:	90dB w/m
Power Handling:	70 / 140 Watts max.
Series:	Two Way

PCH-694T2



Diameter:	6" x 9"
Frequency Range:	65-20,000 Hz
Impedance:	4Ω
Sensitivity:	94dB w/m
Power Handling:	80 / 160 Watts max.
Series:	Two Way

PUNCH SERIES SEPARATES

FEATURES

Neodymium Magnet (ND4)

- Compact Magnet Size
- Increased Gap Strength Over Ferrite

Ferrofluid (ND4, SP34)

- Increased High Temperature
- Optimal Heat Transfer

Kevlar™ Reinforced Spruce Pulp Cones

PVA Treated Cones

Dual Laminated Foam Surround

Vented Pole Piece

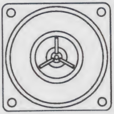
SPL 34/38 - 134/138

- 4" x 6" Plate
- TX Crossover and Midrange High-Pass Crossover
- Biamp Capabilities
- FerroFluid in SP34 Midrange

PUNCH SERIES SEPARATES

SPECIFICATIONS

SPT 4/8

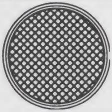


Diameter:	10mm
Frequency Range:	3kHz-20kHz
Impedance:	4Ω or 8Ω
Sensitivity:	91dB w/m
Power Handling:	40 / 80 Watts max.
Series:	Tweeter

SD 4/8

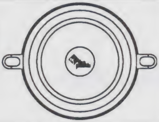
Diameter:	4" x 6" / 3-1/2" plate
Frequency Range:	3kHz-20kHz
Impedance:	4Ω or 8Ω
Sensitivity:	89dB w/m
Power Handling:	50 / 100 Watts max.
Series:	Tweeter

ND 4/8



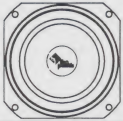
Diameter:	1"
Frequency Range:	3kHz-20kHz
Impedance:	4Ω or 8Ω
Sensitivity:	91dB w/m
Power Handling:	50 / 100 Watts max.
Series:	Tweeter

SP-34/38



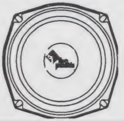
Diameter:	3.5"
Frequency Range:	205Hz-20kHz
Impedance:	4Ω or 8Ω
Sensitivity:	86dB w/m
Power Handling:	50 / 100 Watts max.
Series:	Midrange

SP-44/48



Diameter:	4"
Frequency Range:	85Hz-12kHz
Impedance:	4Ω or 8Ω
Sensitivity:	88dB w/m
Power Handling:	50 / 100 Watts max.
Series:	Midrange

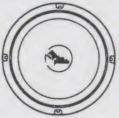
SP-54/58



Diameter:	5.25"
Frequency Range:	96Hz-6,200Hz
Impedance:	4Ω or 8Ω
Sensitivity:	90dB w/m
Power Handling:	80 / 160 Watts max.
Series:	Midrange

PUNCH SERIES SEPARATES

SPECIFICATIONS

SP-64/68 	Diameter: 6.5" Frequency Range: 44Hz-4kHz Impedance: 4Ω or 8Ω Sensitivity: 89dB w/m Power Handling: 80 / 160 Watts max. Series: Woofer
SPL-34/38	Diameter: 4" x 6" plate Frequency Range: 205Hz-2kHz Impedance: 4Ω or 8Ω Sensitivity: 91dB w/m Power Handling: 50 / 100 Watts max. Series: Two-Way
SPL-134/138	Diameter: 4" x 6" plate Frequency Range: 205Hz-2kHz Impedance: 4Ω or 8Ω Sensitivity: 89dB w/m Power Handling: 50 / 100 Watts max. Series: Two-Way

AUDIOPHILE SERIES SEPARATES

FEATURES

Rigid Paper Cone

- ☞ Extended High Frequency Response

Nitrile Rubber Surround

Black Powder Painted CRS Baskets

Flat Spider Mechanism

Two Layer Voice Coil

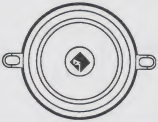
Vented Pole Piece

- ☞ Increased Power Handling
- ☞ Better Voice Coil Heat Dissipation

AUDIOPHILE SERIES SEPARATES

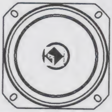
SPECIFICATIONS

RFA-34



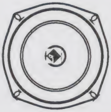
Diameter:	3.5"
Frequency Range:	100Hz-8kHz
Impedance:	4Ω
Sensitivity:	88dB w/m
Power Handling:	50 / 100 Watts
Series:	Midrange

RFA-44



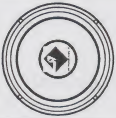
Diameter:	4"
Frequency Range:	100Hz-8kHz
Impedance:	4Ω
Sensitivity:	88dB w/m
Power Handling:	50 / 100 Watts
Series:	Midrange

RFA-54



Diameter:	5.25"
Frequency Range:	45Hz-8kHz
Impedance:	4Ω
Sensitivity:	88dB w/m
Power Handling:	50 / 100 Watts
Series:	Midrange

RFA-64



Diameter:	6.5"
Frequency Range:	45Hz-8kHz
Impedance:	4Ω
Sensitivity:	89dB w/m
Power Handling:	80 / 160 Watts
Series:	Mid-Woofer

AUDIOPHILE SERIES SYSTEMS

FEATURES

Rigid Paper Cone

- ☞ Extended High Frequency Response

Nitrile Rubber Surround

Black Powder Painted CRS Baskets

Flat Spider Mechanism

Two Layer Voice Coil

Vented Pole Piece

- ☞ Increased Power Handling
- ☞ Better Voice Coil Heat Dissipation

ND-4 Flush Mount Assemble

2X-4 Crossover

- ☞ 12dB/octave at 4kHz Linkwitz-Riley Alignment
- ☞ Better Voice Coil Heat Dissipation

AUDIOPHILE SERIES SYSTEMS

SPECIFICATIONS

RFA-414	Diameter: 4" Frequency Range: 94Hz-20kHz Impedance: 4Ω Sensitivity: 86dB w/m Power Handling: 30 / 60 Watts Series: System
RFA-514	Diameter: 5.25" Frequency Range: 44Hz-20kHz Impedance: 4Ω Sensitivity: 86dB w/m Power Handling: 50 / 100 Watts Series: System
RFA-614	Diameter: 6.5" Frequency Range: 39Hz-20kHz Impedance: 4Ω Sensitivity: 89dB w/m Power Handling: 80 / 160 Watts Series: System

POWER SERIES SEPARATES

FEATURES

WOVEN KEVLAR™ CONE

THERMOPLASTIC RUBBER SURROUND

OVERSIZED MAGNET

FLAT SPIDER

FERROFLUID FILLED IN VOICE COIL GAP

MANIFOLD VENTING

LARGER VOICE COILS

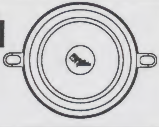
- ☞ 1" (PWR-34/44), 1.5" (PWR-54/64)

VENTED POLE PIECE

POWER SERIES SEPARATES

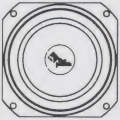
SPECIFICATIONS

PWR-34/38



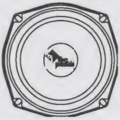
Diameter:	3.5"
Frequency Range:	155Hz-10kHz
Impedance:	4Ω or 8Ω
Sensitivity:	84dB w/m
Power Handling:	70 / 140 Watts max.
Series:	Midrange

PWR-44/48



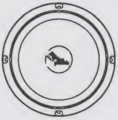
Diameter:	4"
Frequency Range:	85Hz-8kHz
Impedance:	4Ω or 8Ω
Sensitivity:	91dB w/m
Power Handling:	70 / 140 Watts max.
Series:	Midrange

PWR-54/58



Diameter:	5.25"
Frequency Range:	50Hz-6kHz
Impedance:	4Ω or 8Ω
Sensitivity:	90dB w/m
Power Handling:	80 / 160 Watts max.
Series:	Midrange

PWR-64/68



Diameter:	6.5"
Frequency Range:	60Hz-6kHz
Impedance:	4Ω or 8Ω
Sensitivity:	92dB w/m
Power Handling:	80 / 160 Watts max.
Series:	Woofer

Chapter 4

Woofers

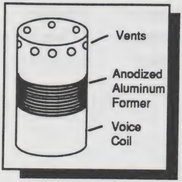
Vented Voice Coil Former

This allows air to flow on both sides of the voice coil with less distortion at lower frequencies.

Feature Vented Voice Coil Former

Function To evenly dissipate heat from the voice coil

Benefit Increased efficiency and power handling with less distortion



Anodized Aluminum Former

The physical support for the voice coil.

Feature Aluminum former

Function To transfer heat away from the voice coil

Benefit Lower operating temperature and less impedance create increased power output

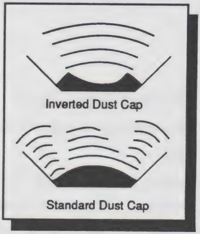
Inverted Dust Cap

To seal the voice coil in order to cool it and decrease cone resonances.

Feature Inverted Dust Cap

Function To assist in cooling the voice coil and decrease reflection

Benefit Lower operating temperature and increased power output



Cast Aluminum Frame

Designed to support the cone and the magnet assembly of a loudspeaker.

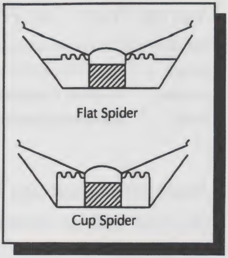
Feature Cast Aluminum Frame

Function To reinforce the cone of the driver and suspension

Benefit Less flex and resonance introduced into the drive

Flat Spider

The flat spider is designed to ensure proper cone control at the former.



Feature	Flat Spider
Function	To keep voice coil centered in the magnetic gap
Benefit	Increased linearity and power handling

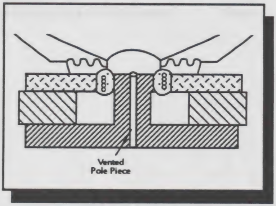
Kevlar™ Reinforced Spruce Pulp Cones

A speaker coating that strengthens the cones.

Feature	Kevlar
Function	Strengthen the cone for a linear response
Benefit	Less cone distortion and strong, low frequency response

Vented Pole Piece

Air ventilation through the former for cooling of the voice coil.



Feature	Vented Pole Piece
Function	To assist in cooling the former and voice coil
Benefit	Better power handling with less resistance introduced from heat.

Dual Laminated Foam Surround

Assists in the suspension of the cone for linear movement.

Feature	Dual Laminated Foam Surround
Function	To control the woofer's excursion with lower surround resonances
Benefit	Less cancellation for increased sensitivity and longer surround life

Nitrile Rubber Surround

Offers increased excursion with consistent control of the cone.

Feature	Nitrile Rubber Surround
Function	To control the movement of the cone with lower edge resonances
Benefit	Flat response and increased excursion

PVA Treated Cones

The PolyVinyl Emulsion Treatment on cones increases the performance with less weight to the cone.

Feature	PVA
Function	Strengthens the cone with less cone flex
Benefit	Flat frequency response

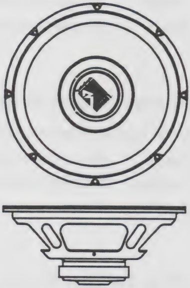
4" Voice Coils

Designed to increase the power handling because the increase of coil windings and heat transfer.

Feature	4" Voice Coil
Function	Conduct power through the coil winding in order to excite the magnetic field
Benefit	Increased power handling and lower impedance rise

SERIES 1 WOOFERS

FEATURES



CRS Basket

- ☞ Heavy Steel for Rigidity
- ☞ Anti-Corrosion Paint (electrostatically applied, baked on)
- ☞ Steel Resonants at Higher Frequency than Die Cast Frame

PVA Treated Cone

Anodized Aluminum, Vented Voice Coil Former

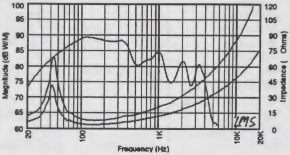
One Piece Stepped Back Plate

- ☞ Allows Extended Voice Coil Excursion
- ☞ Voice Coil Can't Bottom Out on Backplate

SERIES 1 WOOFERS

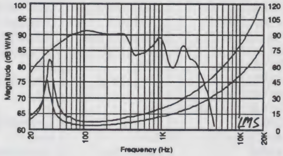
SPECIFICATIONS & LMS DRAWINGS

8" WOOFER	**S1-84**	**S1-88**
POWER:	100 / 200Watts	100 / 200Watts
SENSITIVITY:	89dB w/m	89dB w/m
DIAMETER:	8" (203.2mm)	8" (203.2mm)
MAGNET:	18 oz. (511g)	18 oz. (511g)
VOICE COIL:	1.5"	1.5"
FREQUENCY:	40Hz-1,200Hz	42Hz-1,200Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.32	.33
VAS:	1.059ft^3 (30L)	1.059ft^3 (30L)
RESONANCE:	40Hz	42Hz
XMAX	.19" (4.7mm)	.19" (4.7mm)



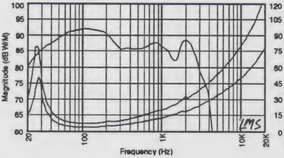
Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

10" WOOFER	**S1-104**	**S1-108**
POWER:	100 / 200 Watts	100 / 200 Watts
SENSITIVITY:	91dB w/m	91dB w/m
DIAMETER:	10" (254.0mm)	10" (254.0mm)
MAGNET:	18 oz. (511g)	18 oz. (511g)
VOICE COIL:	1.5"	1.5"
FREQUENCY:	36Hz-2kHz	38Hz-2kHz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.36	.38
VAS:	2.401ft^3 (68L)	2.401ft^3 (68L)
RESONANCE:	36Hz	38Hz
XMAX	.20" (5.2mm)	.20" (5.2mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

12" WOOFER	**S1-124**	**S1-128**
POWER:	100 / 200 Watts	100 / 200 Watts
SENSITIVITY:	92dB w/m	92dB w/m
DIAMETER:	12" (304.8mm)	12" (304.8mm)
MAGNET:	28 oz. (795.2g)	28 oz. (795.2g)
VOICE COIL:	1.5"	1.5"
FREQUENCY:	28Hz-2,500Hz	29-2,500Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.33	.41
VAS:	6.569ft^3 (186L)	6.569ft^3 (186L)
RESONANCE:	28Hz	29Hz
XMAX	.20" (5.2mm)	.20" (5.2mm)

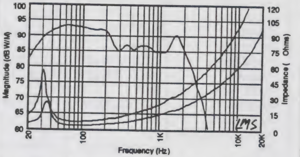


Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

SERIES 1 WOOFERS

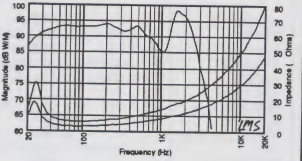
SPECIFICATIONS & LMS DRAWINGS

15" WOOFER	S1-154	S1-158
POWER:	150 / 300 Watts	150 / 300 Watts
SENSITIVITY:	93dB w/m	93dB w/m
DIAMETER:	15" (381.0mm)	15" (381.0mm)
MAGNET:	38 oz. (1079.2g)	38 oz. (1079.2g)
VOICE COIL:	2.0"	2.0"
FREQUENCY:	26Hz-2,500Hz	27Hz-2,500Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.44	.51
VAS:	11.336ft^3 (321L)	11.336ft^3 (321L)
RESONANCE:	26Hz	27Hz
XMAX	.23" (5.9mm)	.23" (5.9mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

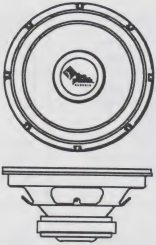
18" WOOFER	S1-184	S1-188
POWER:	200 / 400 Watts	200 / 400 Watts
SENSITIVITY:	92dB w/m	92dB w/m
DIAMETER:	18" (457.2mm)	18" (457.2mm)
MAGNET:	64 oz. (1817.6g)	64 oz. (1817.6g)
VOICE COIL:	2.5"	2.5"
FREQUENCY:	25Hz-2kHz	26Hz-2kHz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.57	.80
VAS:	13.985ft^3 (392L)	13.985ft^3 (392L)
RESONANCE:	25Hz	26Hz
XMAX	.31" (8.0mm)	.31" (8.0mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

PUNCH CLASSIC WOOFERS

FEATURES



KEVLAR™ REINFORCED SPRUCE PULP CONE (KSP)

INVERTED DUST CAP

LARGE STRONTIUM FERRITE MAGNET

- High Power Handling
- 12-15-18 Use Stacked Magnets for Improved Sensitivity

ANODIZED ALUMINUM VOICE COIL FORMER

VENTED POLE PIECES

DUAL LAMINATE SURROUND

CHAMFERED FIELD PLATE

- Top Field Plate Has Bevel to Create Focused Magnetic Image on Voice Coil

ONE PIECE STEPPED BACK PLATE

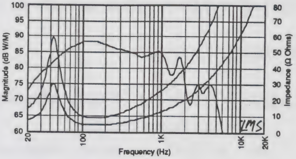
- Voice Coil Can't Bottom Out
- Allows Extended Voice Coil Excursion

FLAT SPIDER

PUNCH CLASSIC WOOFERS

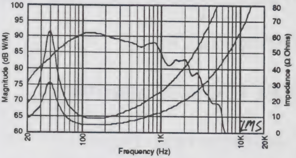
SPECIFICATIONS & LMS DRAWINGS

8" WOOFER	PCH-408	PCH-808
POWER:	150 / 300 Watts	150 / 300 Watts
SENSITIVITY:	89dB w/m	89dB w/m
DIAMETER:	8" (203.2mm)	8" (203.2mm)
MAGNET:	30 oz. (850g)	30 oz. (850g)
VOICE COIL:	2"	2"
FREQUENCY:	42Hz-1,500Hz	45Hz-1,500Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.34	.34
VAS:	.795ft^3 (22.5L)	.795ft^3 (22.5L)
RESONANCE:	42Hz	45Hz
XMAX	.26" (6.7mm)	.26" (6.7mm)



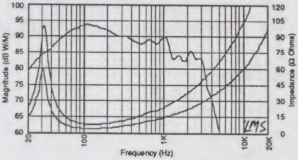
Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

10" WOOFER	PCH-410	PCH-810
POWER:	150 / 300 Watts	150 / 300 Watts
SENSITIVITY:	91dB w/m	91dB w/m
DIAMETER:	10" (254mm)	10" (254mm)
MAGNET:	38 oz. (1077g)	38 oz. (1077g)
VOICE COIL:	2"	2"
FREQUENCY:	38Hz-1,500	40Hz-1,500Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.34	.39
VAS:	1.7ft^2 (47L)	1.7ft^2 (47L)
RESONANCE:	38Hz	40Hz
XMAX	.26" (6.7mm)	.26" (6.7mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

12" WOOFER	PCH-124	PCH-128
POWER:	200 / 400 Watts	200 / 400 Watts
SENSITIVITY:	94dB w/m	94dB w/m
DIAMETER:	12" (305mm)	12" (305mm)
MAGNET:	60 oz. (1701g)	60 oz. (1701g)
VOICE COIL:	2"	2"
FREQUENCY:	32Hz-250Hz	32Hz-250Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.30	.32
VAS:	4.697 ft^3 (133L)	4.697ft^3 (133L)
RESONANCE:	32Hz	33Hz
XMAX	.19" (4.7mm)	.19" (4.7mm)

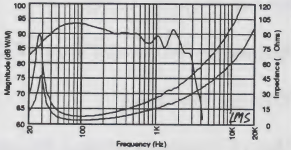


Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

PUNCH CLASSIC WOOFERS

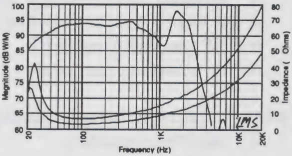
SPECIFICATIONS & LMS DRAWINGS

15" WOOFER	PCH-154	PCH-158
POWER:	200 / 400 Watts	200 / 400 Watts
SENSITIVITY:	94dB w/m	94dB w/m
DIAMETER:	15" (381mm)	15" (381mm)
MAGNET:	60 oz. (1701g)	60 oz. (1701g)
VOICE COIL:	2"	2"
FREQUENCY:	28Hz-250Hz	28Hz-250Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.40	.42
VAS:	10.347 ft^3 (293L)	10.347ft^3 (293L)
RESONANCE:	28Hz	29Hz
XMAX	.19" (4.7mm)	.19" (4.7mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

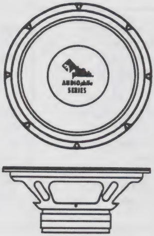
18" WOOFER	PCH-184	PCH-188
POWER:	250 / 500 Watts	250 / 500 Watts
SENSITIVITY:	94dB w/m	94dB w/m
DIAMETER:	18" (457mm)	18" (457mm)
MAGNET:	105 oz. (2977g)	105 oz. (2977g)
VOIC COIL:	2.5"	2.5"
FREQUENCY:	23Hz-200Hz	23Hz-200Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.41	.53
VAS:	17.975 ft^3 (509L)	17.975ft^3 (509L)
RESONANCE:	23Hz	24Hz
XMAX	.32" (8.1mm)	.32" (8.1mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

AUDIOPHILE SERIES WOOFERS

FEATURES



Kevlar™ (KSP) Cone

Anodized Aluminum Voice Coil Former

Vented Pole Piece

High Compliant Nitrile Rubber Surround

Longer Voice Coil

- Increased Linear Excursion (XMax)
- 8" and 10" use a 2" Voice Coil
- 12" Use a 2.5" Voice Coil
- Increases Heat Dissipation

Large Strontium Ferrite Magnet

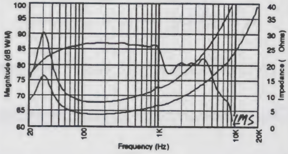
- High Power Handling
- 10-12 Use Stacked Magnets for Improved Sensitivity

Flat Spiders

AUDIOPHILE SERIES WOOFERS

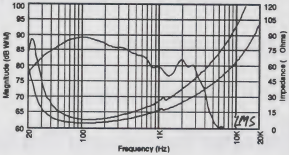
SPECIFICATIONS & LMS DRAWINGS

8" WOOFER	RFA-84	RFA-88
POWER:	150 / 300 Watts	150 / 300 Watts
SENSITIVITY:	86dB w/m	86dB w/m
DIAMETER:	8" (203.2mm)	8" (203.2mm)
MAGNET:	22 oz. (623.7g)	22 oz. (623.7g)
VOICE COIL:	2"	2"
FREQUENCY:	30Hz-1kz	30Hz-1kHz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	0.42	0.52
VAS:	2.154 cu. ft. (61L)	2.154 cu. ft. (61L)
RESONANCE:	29Hz	31Hz
XMAX	.25" (6.3mm)	.25" (6.3mm)



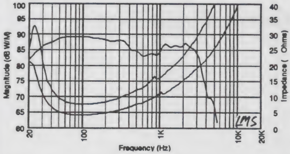
Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

10" WOOFER	RFA-104	RFA-108
POWER:	150 / 300 Watts	150 / 300 Watts
SENSITIVITY:	88dB w/m	88dB w/m
DIAMETER:	10" (254.0mm)	10" (254.0mm)
MAGNET:	60 oz. (1,701g)	60 oz. (1,701g)
VOICE COIL:	2"	2"
FREQUENCY:	20Hz-1kHz	20Hz-1kHz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	0.21	0.23
VAS:	4.026 cu. ft. (114L)	4.026 cu. ft. (114L)
RESONANCE:	21Hz	22Hz
XMAX	.19" (4.7mm)	.19" (4.7mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

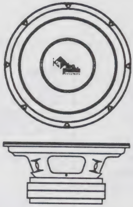
12" WOOFER	RFA-124	RFA-128
POWER:	250 / 500 Watts	250 / 500 Watts
SENSITIVITY:	89dB w/m	89dB w/m
DIAMETER:	12" (304.8mm)	12" (304.8mm)
MAGNET:	64 oz. (1,814.4g)	64 oz. (1,814.4g)
VOICE COIL:	2.5"	2.5"
FREQUENCY:	20Hz-1kHz	20Hz-1kHz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	0.38	0.48
VAS:	9.182 cu. ft. (260L)	9.182 cu. ft. (260L)
RESONANCE:	21Hz	22Hz
XMAX	.37" (9.3mm)	.37" (9.3mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

POWER SERIES WOOFERS

FEATURES



Anodized Aluminum Voice Coil Former

- ☞ CRS basket on 8"
- ☞ Woven Kevlar® cone on 8"

Vented Pole Piece with Aerodynamic Shape

Dual Laminated Foam Surrounds

Three Inch and Four Inch Voice Coil

- ☞ Power 10" has a 3" Voice Coil
- ☞ Power 12" through 18" have a 4" Voice Coil

Cast Aluminum Baskets

Gold Plated Binding Posts

Insulated Silver-Plated Flex Leads

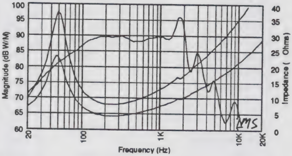
Flat Spiders

Magnet Rubber Boot

POWER SERIES WOOFERS

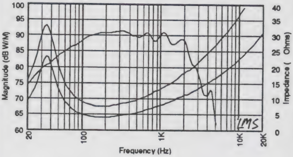
SPECIFICATIONS & LMS DRAWINGS

8" WOOFER	**PWR-408**	**PWR-808**
POWER:	200 / 400 Watts	200 / 400 Watts
SENSITIVITY:	90dB w/m	90dB w/m
DIAMETER:	8" (203.2mm)	8" (203.2mm)
MAGNET:	64 oz. (1814.4g)	64 oz. (1814.4g)
VOICE COIL:	2.5"	2.5"
FREQUENCY:	40Hz-2,500Hz	42Hz-2,500Hz
IMPEDANCE:	4 Ω	8 Ω
TOTAL Q FACTOR:	0.27	0.32
VAS:	1.0 ft.^3 (30L)	1.0 ft.^3 (30L)
RESONANCE:	40Hz	42Hz
XMAX	.18" (4.8mm)	.18" (4.8mm)



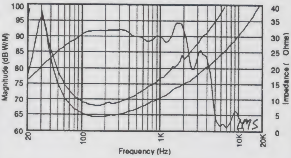
Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

10" WOOFER	**PWR-410**	**PWR-810**
POWER:	300 / 600 Watts	200 / 400 Watts
SENSITIVITY:	91dB w/m	91dB w/m
DIAMETER:	10" (254.0mm)	10" (254.0mm)
MAGNET:	102 oz. (2892g)	102 oz. (2892g)
VOICE COIL:	3"	3"
FREQUENCY:	30Hz-2,500Hz	33Hz-2,500Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	.22	.28
VAS:	2.5 ft.^3 (71L)	2.5 ft.^3 (71L)
RESONANCE:	30Hz	33Hz
XMAX	.30" (7.7mm)	.30" (7.7mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

12" WOOFER	**PWR-412**	**PWR-812**
POWER:	600 / 1200 Watts	600 / 1200 Watts
SENSITIVITY:	91dB w/m	91dB w/m
DIAMETER:	12" (304.8mm)	12" (304.8mm)
MAGNET:	130 oz. (3685g)	130 oz. (3685g)
VOICE COIL:	4"	4"
FREQUENCY:	25Hz-1,500Hz	27Hz-1,500Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	0.26	0.31
VAS:	4.6 ft.^3 (133L)	4.6 ft.^3 (133L)
RESONANCE:	25Hz	27Hz
XMAX	.24" (6.2mm)	.24" (6.2mm)

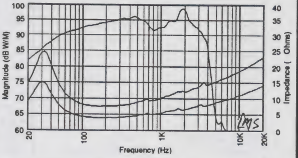


Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

POWER SERIES WOOFERS

SPECIFICATIONS & LMS DRAWINGS

15" WOOFER	**PWR-415**	**PWR-815**
POWER:	600 / 1200 Watts	600 / 1200 Watts
SENSITIVITY:	94dB w/m	94dB w/m
DIAMETER:	15" (381.0mm)	15" (381.0mm)
MAGNET:	116 oz. (3288g)	116 oz. (3288g)
VOICE COIL:	4"	4"
FREQUENCY:	24Hz-250Hz	25Hz-250Hz
IMPEDANCE:	4Ω	8Ω
TOTAL Q FACTOR:	0.28	0.31
VAS:	11 ft.^3 (334L)	11 ft.^3 (334L)
RESONANCE:	24Hz	25Hz
XMAX:	.24" (6.2mm)	.24" (6.2mm)



Upper impedance curve is for 8 Ohm, lower is for 4 Ohm.

SERIES 1 BANDPASS WOOFERS

FEATURES

FOURTH ORDER BANDPASS DESIGN

- Natural High Frequency Cutoff
- 12dB Per Octave Slope

OCTAGON SHAPED TUBE ENCLOSURES

- Durable Carpet Wrap Finish
- Less Standing Waves
- Rounded Port Design
- Increased Efficiency

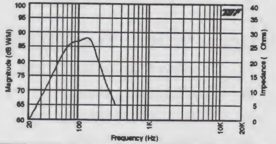
HIGH POWER HANDLING

SERIES 1 BANDPASS WOOFERS

SPECIFICATIONS

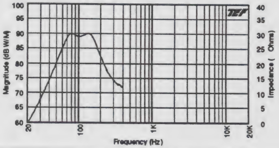
BP-600

Diameter:	6"
Frequency Range:	60-135 Hz
Impedance:	4Ω
Sensitivity:	88dB w/m
Power Handling:	100 Watts



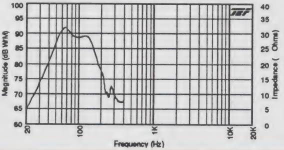
BP-690

Diameter:	6" x 9"
Frequency Range:	56-118 Hz
Impedance:	4Ω
Sensitivity:	90dB w/m
Power Handling:	150 Watts



BP-1000

Diameter:	10"
Frequency Range:	54-126 Hz
Impedance:	4Ω
Sensitivity:	92dB w/m
Power Handling:	150 Watts



THE BOX THAT ROCKS

FEATURES

Latest In Full Range Speaker Enclosures

- Custom 4 Ohm Drivers
- Tuned Bass Reflex Cabinets
- Designed For Trucks And Hatchbacks
- Cabinets Built From Particle Board

2" Ports

- For Faster And More Efficient Bass Exhaust
- Rounded Port Ends

Black Metal Grilles

High Power Handling

93dB Sensitive

Vented Voice Coils

THE BOX THAT ROCKS

SPECIFICATIONS

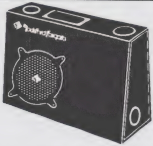
BTR-82

Diameter:	8"
Impedance:	4Ω
Sensitivity:	93dB w/m
Power Handling:	200 Watts
Series:	Two Way

BTR-83

Diameter:	8"
Impedance:	4Ω
Sensitivity:	93dB w/m
Power Handling:	200 Watts
Series:	Three Way

BTR-103



Diameter:	10"
Impedance:	4Ω
Sensitivity:	93dB w/m
Power Handling:	200 Watts
Series:	Three Way

BTR-123

Diameter:	12"
Impedance:	4Ω
Sensitivity:	93dB w/m
Power Handling:	350 Watts
Series:	Three Way

BTR-153

Diameter:	15"
Impedance:	4Ω
Sensitivity:	93dB w/m
Power Handling:	350 Watts
Series:	Three Way

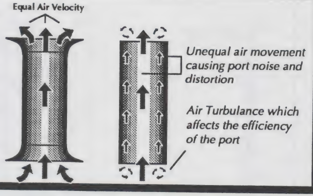
AEROPORT TUBE

3" AND 4" DIAMETERS

15" TUBE LENGTH

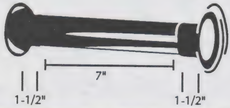
3 PIECE ASSEMBLY

Impedance matches the air inside the enclosure with the port to the air outside increasing efficiency which increases power handling and SPL with less distortion enabling a 10% air velocity which is a figure given in Term Pro.



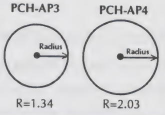
Calculating Port Length

Example: 10" Total Usable Port Length



After the enclosure has been designed, the following formula may be useful when calculating the port length:

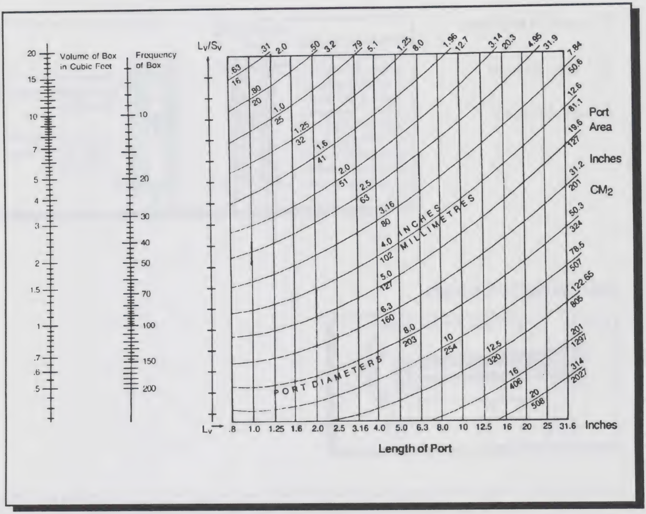
$$Lv = \frac{8466.4R^2}{(Vb)Fb^2} - 1.463R$$



Lv = port length in inches
Vb = volume of box in cubic feet
Fb = tuning frequency
R = radius of port

When cutting the Aeroport center tube, the length should be cut 3" shorter than calculated.

NOMOGRAM



Using The Port Tuning Nomogram

Use this chart by drawing a line connecting the box volume and the desired port frequency, and extending it through the third vertical column, labeled Lv/Sv. From the third vertical column, draw a horizontal line across the page. The curved lines that your horizontal line intersect indicate the diameter of ports that can be used (in inches and centimeters). Areas are given at the margin for use with rectangular ports. To get the port length, draw a vertical line from the place your horizontal line intersects the curved line for the diameter, to the bottom of the chart where the length shown. Add 10% to the port length the chart gives. Boxes always act smaller than third measured volume would indicate.

Chapter 5

Signal Processors

Zobel Network

Impedance compensation circuit which compensates for the inductance rise of a voice coil.

Feature	Zobel Network
Function	Smooth the impedance of the driver at 10kHz
Benefit	Reduces harshness caused by the inductance rise of the voice coil

Series Notch Filter

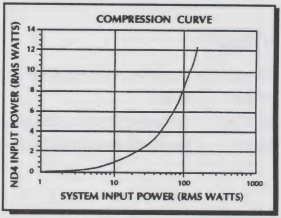
This is centered at 1.5kHz to limit the peak in the impedance curve due to tweeter resonance.

Feature	Series Notch Filter
Function	To control the impedance rise of the tweeter
Benefit	A linear response with a tweeter and midrange on the same vertical plane

Optical Compression Circuit

This will apply "soft limiting" of the input power to the tweeter

Feature	Optical Compression Circuit
Function	To limit power to the tweeter at the beginning threshold 14 volts RMS (50 Watts) to maximum compression at 20 Volts RMS (100 Watts)
Benefit	Undetectable and extends the life of the tweeter



Iron Core Coils

Use of a magnetic material in conjunction with the turns of wire to obtain more inductance per turn.

Feature	Iron Core Coils
Function	To pass only the frequencies that its value will allow
Benefit	A lower tolerance compared to air core for better tracking of the crossover point

Mylar Capacitors

Designed for low tolerance and high power handling for increased capacitive reactance.

Feature Mylar Capacitors

Function To increase resistance to low frequencies

Benefit Increased tracking ability of the crossover point so there is less frequency interaction with a tweeter and midrange

Global

To store certain EQ levels and crossover points and preamp settings and implement them at a given volume level.

Feature Global

Function Memory storage of four presets of crossover equalization and preamp settings

Benefit Ability to adjust multiple functions with the use of the volume control

VDISC

The ability to toggle between two global presets based on the volume's position.

Feature VDISC

Function To interact with the global setting and activate preset parameters at a particular volume level

Benefit The use of one control to operate equalization, crossover and preamp in real time

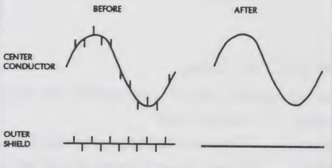
Differential Inputs

This circuit will amplify the difference between two conductors or wires in the preamp stage

Feature Differential Inputs

Function To prevent noise induced into the audio path

Benefit Noise spikes are not amplified they are cancelled out of the system



Differential Inputs only amplify the difference between two conductors. Note the noise spikes appear on both the center conductor and the outer shield and therefore are not amplified.

Linkwitz-Riley

A crossover alignment with a roll off response Q of .49 which permits more flexibility in driver placement.

Feature	Linkwitz-Riley
Function	To lower the sensitivity to the offset of a tweeter and midrange mounted on the same vertical axis
Benefit	Equal group delay between the tweeter and midrange resulting in a smooth response

Butterworth

A crossover alignment with a roll off response Q of .7 which yields the flattest response at the expense of some damping

Feature	Butterworth
Function	To give an extended flat response
Benefit	No peaks in response

Bessel

A crossover alignment with a roll off response Q of .58 which will not sum flat the response.

Feature	Bessel
Function	Increased magnitude in the sum of the low and high-pass crossover points
Benefit	Tweeter does not have to be mounted on the same vertical axis of the midrange

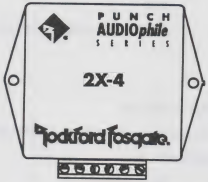
Chebychev

A crossover alignment with a roll off response Q of 1 which, when used with a woofer, will usually get an increased response.

Feature	Chebychev
Function	To give a 6dB increase at the summed crossover point
Benefit	Increased output at a particular frequency without the use of equalization

PUNCH AUDIOPHILE PASSIVE CROSSOVERS

2X-4



LINKWITZ-RILEY ALIGNMENT

12dB/OCTAVE AT 4KHZ

SERIES NOTCH FILTER

ZOBEL IMPEDANCE COMPENSATING CIRCUIT

OPTICAL COMPRESSION CIRCUIT

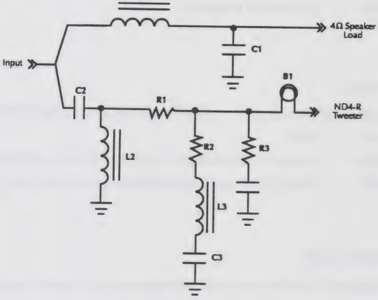
HIGH-PASS RESISTOR PAD

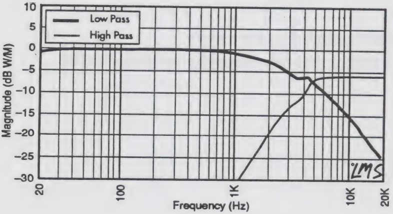
12 GAUGE TERMINAL CONNECTORS

MYLAR CAPACITORS

IRON CORE COILS

PACKAGED WITH 414, 514, 614





PUNCH AUDIOPHILE PASSIVE CROSSOVERS

LP-1/LP-2

Iron Core Coils 3% Tolerance

6dB/Octave Butterworth Alignment

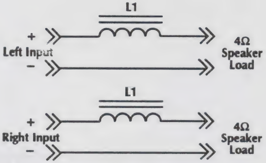
Multiple Crossover Points

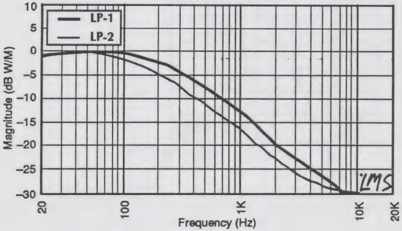
Stereo or Mono Input

12 Gauge Terminal Connector

Coils Wired Series In Mono Mode

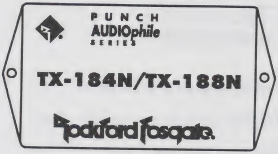
	STEREO			MONO		
	2Ω	4Ω	8Ω	2Ω	4Ω	8Ω
LP-1	100Hz	200Hz	400Hz	50Hz	100Hz	200Hz
LP-2	50Hz	100Hz	200Hz	25Hz	50Hz	100Hz





PUNCH AUDIOPHILE PASSIVE CROSSOVERS

184N/188N



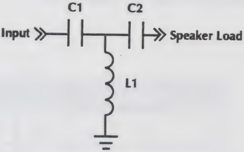
Butterworth

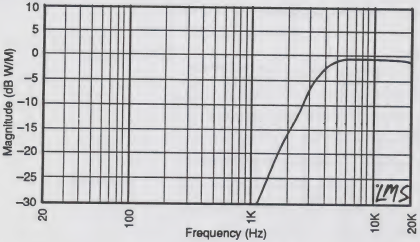
18dB/Octave at 4kHz

Polarized Faston Terminals

Mylar Capacitors

Included with ND-4 Tweeter





PUNCH ELECTRONIC CROSSOVERS

- **The Crossover Module programmed with a Butterworth Alignment can be used High-Pass or Low-Pass without changing any resistor values.**
- **The Crossover Cards programmed with a Bessel, Linkwitz-Riley and Chebychev will be programmed specifically for High-Pass or Low-Pass.**

Bessel Alignment

Use 5% resistors in conjunction with .022µF standard capacitor.

	Low Pass		High Pass	
Freq.	R1	R2	R1	R2
30Hz	200k Ω	100k Ω	470k Ω	360k Ω
35Hz	180k Ω	82k Ω	390k Ω	300k Ω
40Hz	150k Ω	68k Ω	330k Ω	270k Ω
50Hz	120k Ω	56k Ω	270k Ω	220k Ω
60Hz	100k Ω	47k Ω	240k Ω	280k Ω
75Hz	82k Ω	39k Ω	200k Ω	160k Ω
90Hz	68k Ω	33k Ω	160k Ω	130k Ω
106Hz	56k Ω	27k Ω	130k Ω	100k Ω
130Hz	47k Ω	22k Ω	110k Ω	82k Ω
160Hz	39k Ω	18k Ω	91k Ω	75k Ω
190Hz	30k Ω	15k Ω	75k Ω	62k Ω
250Hz	24k Ω	12k Ω	56k Ω	43k Ω
300Hz	20k Ω	10k Ω	47k Ω	36k Ω
350Hz	18k Ω	8.2k Ω	39k Ω	30k Ω
425Hz	15k Ω	6.8k Ω	33k Ω	27k Ω
520Hz	12k Ω	5.6k Ω	27k Ω	22k Ω
610Hz	10k Ω	4.7k Ω	24k Ω	18k Ω
740Hz	8.2k Ω	3.9k Ω	20k Ω	16k Ω
870Hz	6.8k Ω	3.3k Ω	16k Ω	13k Ω
1.1kHz	5.6k Ω	2.7k Ω	13k Ω	10k Ω
1.3kHz	4.7k Ω	2.2k Ω	11k Ω	8.2k Ω
1.6kHz	3.9k Ω	1.8k Ω	9.1k Ω	7.5k Ω
1.9kHz	3.0k Ω	1.5k Ω	7.5k Ω	6.2k Ω
2.5kHz	2.4k Ω	1.2k Ω	5.6k Ω	4.3k Ω
3.0kHz	2.0k Ω	1.0k Ω	4.7k Ω	3.6k Ω
3.5kHz	1.8k Ω	820 Ω	3.9k Ω	3.0k Ω
4.25kHz	1.5k Ω	680 Ω	3.3k Ω	2.7k Ω
5.2kHz	1.2k Ω	560 Ω	2.7k Ω	2.2k Ω
6.1kHz	1.0k Ω	470 Ω	2.4k Ω	1.8k Ω
7.4kHz	820 Ω	390 Ω	2.0k Ω	1.6k Ω
8.7kHz	680 Ω	330 Ω	1.6k Ω	1.3k Ω
11kHz	560 Ω	270 Ω	1.3k Ω	1.0k Ω

Butterworth Alignment

Use 5% resistors in conjunction with .022µF standard capacitor.

	Low Pass		High Pass	
Freq.	R1	R2	R1	R2
18.5Hz	390k Ω	390k Ω	390k Ω	390k Ω
26Hz	270k Ω	270k Ω	270k Ω	270k Ω
33Hz	220k Ω	220k Ω	220k Ω	220k Ω
40Hz	180k Ω	180k Ω	180k Ω	180k Ω
48Hz	150k Ω	150k Ω	150k Ω	150k Ω
60Hz	120k Ω	120k Ω	120k Ω	120k Ω
72Hz	100k Ω	100k Ω	100k Ω	100k Ω
88Hz	82k Ω	82k Ω	82k Ω	82k Ω
106Hz	68k Ω	68k Ω	68k Ω	68k Ω
130Hz	56k Ω	56k Ω	56k Ω	56k Ω
154Hz	47k Ω	47k Ω	47k Ω	47k Ω
185Hz	39k Ω	39k Ω	39k Ω	39k Ω
220Hz	33k Ω	33k Ω	33k Ω	33k Ω
270Hz	27k Ω	27k Ω	27k Ω	27k Ω
330Hz	22k Ω	22k Ω	22k Ω	22k Ω
400Hz	18k Ω	18k Ω	18k Ω	18k Ω
480Hz	15k Ω	15k Ω	15k Ω	15k Ω
600Hz	12k Ω	12k Ω	12k Ω	12k Ω
720Hz	10k Ω	10k Ω	10k Ω	10k Ω
880Hz	8.2k Ω	8.2k Ω	8.2k Ω	8.2k Ω
1.06kHz	6.8k Ω	6.8k Ω	6.8k Ω	6.8k Ω
1.3kHz	5.6k Ω	5.6k Ω	5.6k Ω	5.6k Ω
1.54kHz	4.7k Ω	4.7k Ω	4.7k Ω	4.7k Ω
1.85kHz	3.9k Ω	3.9k Ω	3.9k Ω	3.9k Ω
2.2kHz	3.3k Ω	3.3k Ω	3.3k Ω	3.3k Ω
2.7kHz	2.7k Ω	2.7k Ω	2.7k Ω	2.7k Ω
3.3kHz	2.2k Ω	2.2k Ω	2.2k Ω	2.2k Ω
4.0kHz	1.8k Ω	1.8k Ω	1.8k Ω	1.8k Ω
4.8kHz	1.5k Ω	1.5k Ω	1.5k Ω	1.5k Ω
6.0kHz	1.2k Ω	1.2k Ω	1.2k Ω	1.2k Ω
7.2kHz	1.0k Ω	1.0k Ω	1.0k Ω	1.0k Ω
8.8kHz	820 Ω	820 Ω	820 Ω	820 Ω
10.6kHz	680 Ω	680 Ω	680 Ω	680 Ω
13.0kHz	560 Ω	560 Ω	560 Ω	560 Ω
15.4kHz	470 Ω	470 Ω	470 Ω	470 Ω

PUNCH ELECTRONIC CROSSOVERS

2X

3 Year Warranty

12dB/Octave Slope Butterworth Alignment

2-Way Selectable Crossover

2 Channel Input

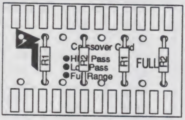
4 Channel Output

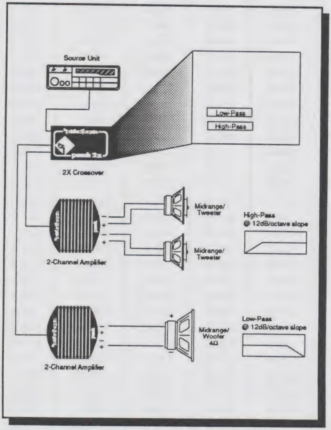
Gold-Plated RCAs

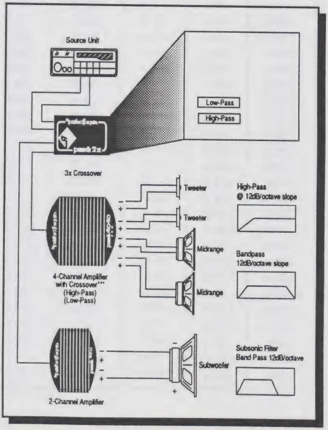
Individually Selectable High-Pass, Low-Pass and Full Range

Unity Gain Output

Butterworth, Bessel, Linkwitz-Riley, Chebychev Alignment Capability

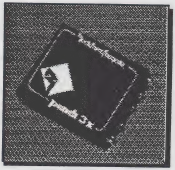






PUNCH ELECTRONIC CROSSOVERS

3X



3 YEAR WARRANTY

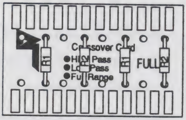
12dB/OCTAVE SLOPE BUTTERWORTH ALIGNMENT

3-WAY SELECTABLE CROSSOVER

4 CHANNEL INPUT

6 CHANNEL OUTPUT

GOLD-PLATED RCAs



INDIVIDUALLY SELECTABLE HIGH-PASS, LOW-PASS AND FULL RANGE

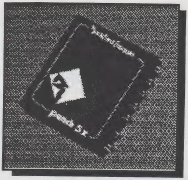
DUAL FILTERED NON-FADED OUTPUT WITH PHASE REVERSAL SWITCH

UNITY GAIN OUTPUT

BUTTERWORTH, BESSEL, LINKWITZ-RILEY, CHEBYCHEV ALIGNMENT CAPABILITY

PUNCH ELECTRONIC CROSSOVERS

5X



3 Year Warranty

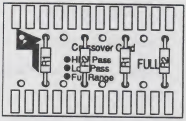
12dB/Octave Slope Butterworth Alignment

5-Way Selectable Crossover

4 Channel Input

6 Channel Output

Gold-Plated RCAs

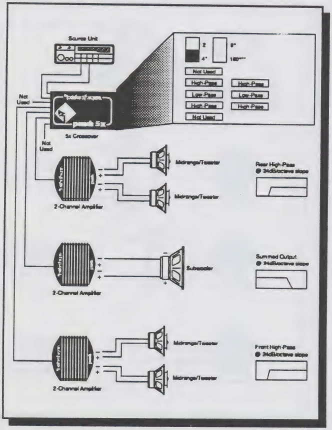


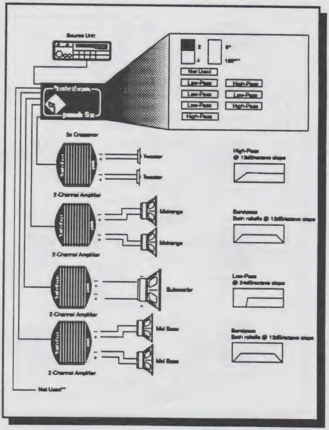
Individually Selectable High-Pass, Low-Pass and Full Range

Dual Filtered Non-Faded Output with Phase Reversal Switch

Unity Gain Output

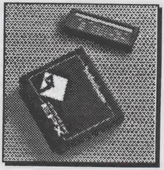
Butterworth, Bessel, Linkwitz-Riley, Chebychev Alignment Capability





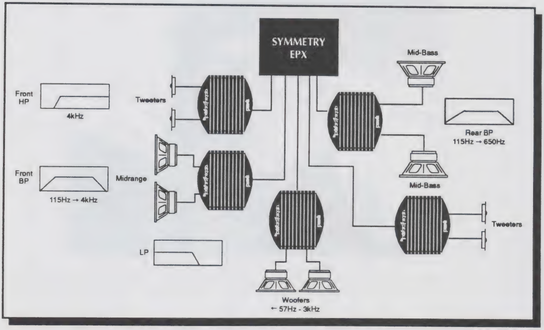
SYMMETRY EPX

EPX



EXCLUSIVE ROCKFORD FEATURE

- 5-WAY 10 CHANNEL CROSSOVER
- 14 BAND STEREO EQUALIZER
  - 4 PRESETS
  - HALF OCTAVE CENTERS: 32, 45, 60, 90, 125, 180, 250, 375 500Hz
  - OCTAVE CENTERS: 1K, 2K, 4K, 8K, 16KHz
  - ±12DB
- PRE-AMP WITH DUAL SOURCE INPUT
- "NEW" 4 GLOBAL PRESETS FOR: EQUALIZER, CROSSOVER, PRE-AMP
- VDISC "VOLUME DEPENDENT INTERACTIVE SYSTEM"
- "NEW" ILLUMINATED RDAT
- 8K BYTES OF NON-VOLATILE STORAGE
- POSITIVE LCD DISPLAY



SYMMETRY

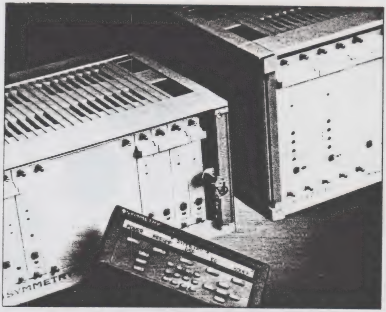
The Concept Of SYMMETRY

SYMMETRY is a modular audio/security/control system designed for maximum versatility. Its modular architecture allows you to easily upgrade your system as new products or technology becomes available.

The basic SYMMETRY system consists of a SYMMETRY Housing, Power Supply Module, Microcontroller Module, Preamplifier Module (PAM) and Missing Link Transmitter (MLT). All are controlled by a hand-held, wired Remote Data Access Terminal (RDAT), also part of the standard package.

The SYM-CSV Controller Housing has a clear plexiglas™ door. Modules slide into runners at the top and bottom of the housing, and are secured with two thumbscrews. Up to twelve modules can be used in one controller housing. Six of the module slots are dedicated to control and interface applications and the other six slots are for audio signal processing chores using analog, digital, or hybrid circuitry. The audio processing cards are classified as preamplifier modules, signal processors, or output modules; however, they must be configured from left to right as follows: outputs, preamplifiers, and signal processors.

A variety of jacks and connectors can be found on the housing's back panel. There is a fifty-pin bus expansion port which can be used to daisy chain multiple controllers together. SYMMETRY can handle a maximum of 256 modules, including up to four of each of sixty-four different module types.



Audio Modules

SYMMETRY's control and audio processing functions are initiated by plug-in modules.

PSM The Power Supply Module converts battery power into audio power. It incorporates five LEDs that represent battery voltage, self-power, positive voltage, negative voltage, and remote turn-on.

CPM The Microcontroller is the brain that has 8k of static RAM, 8k of EPROM and also 8k of EEPROM for non-volatile storage. The module board has additional EPROM sockets to add up to 40k in additional software. There are also four LEDs that indicate power on, receive transmission, transmit transmission, and remote turn-on.

PAM The Preamplifier Module controls volume, muting, balance and fader controls plus bass and treble. The volume can be varied over an 80 dB range. When SYMMETRY is turned on the preamp gradually raises the volume until it reaches the level at which it was set when the system was turned off. You can also use extra preamp modules to create multiple listening areas. The driver can listen to a CD while the passengers watch a video and have the audio run through the system around them. There are two LEDs that indicate power on and when the PAM is being utilized.

MLT The Missing Link Transmitter converts a source unit's unbalanced output to balanced differential lines, which are fully shielded from any R.F.I. The MLT converts up to four unbalanced stereo outputs to a maximum amplified 30 volt signal to the PAM via a shielded multi-pin cable. There is also a selectable ground reference on the inputs to prevent any ground loops.

EQ1 This is a 14-band analog equalization module that is a half octave below 500Hz and one octave above 500Hz. These bands can be boosted or cut by 12dB. Over 4 curves can be stored in memory. The card has two LEDs. One lights up when the system is on, the other when a module is selected.

EQ3 Utilizes a 28 band stereo equalizer that consists of two modules – EQ3L and EQ3H. Together these two modules have center frequencies established with 1/3 octave intervals in the difficult to adjust frequencies below 500 Hz. All the frequency ranges have a narrow "Q" with plus or minus 12dB of adjustment.

DSP The Digital Signal Processing is done in the digital domain. It analyzes the car's acoustical response via a microphone. It then computes an inverse equalization curve. This process is done by defeating any equalization including the tone controls so they won't alter the analysis. It then proceeds to output pink noise and then compares it to the signal that was received by the microphone. After all the data has been received in the DSP module, it then compensates for equalization and time alignment. This is all accomplished through A/D and D/A convertors. This module has five LEDs. The first two are system power on. The next one illuminates when it is analyzing and equalizing the system. The fourth LED blinks when the levels are being set. And the last LED illuminates when it is automatically processing the data.

LTM The Loop Through Module connects the line level processing circuits to the power amplifiers in the amplifier housing. This is not required if you are using the XOM.

XOM The Crossover Module has sixteen analog crossovers and four input channels that extend to the sixteen output channels. This includes front highpass, front bandpass 1, front bandpass 2, front center channel bandpass, rear highpass, rear bandpass 1, rear bandpass 2, and constant bass lowpass. You can set each band from one of 256 preset crossover frequencies and vary the output ±12dB. This whole process can be transmitted through one multi-pin cable; the DB-25. This module has two LEDs. The first one is system power and the other is module selection.

ASV 420 The Amplifier SYMMETRY Vertical housing has four amplifier modules that are rated 20 watts x 4 or 35 watts x 4 into 2Ω. You can also bridge it into two channels for 80 watts per channel. A bridged channel must have a four Ohm load. Each module has a separate left and right input sensitivity. The housing includes a thermostatically controlled fan to cool the amp modules. Each module has LEDs to indicate system power and clipping.

BOB The Break-Out Box is an adaptor box which was designed to utilize our new Punch amplifiers with SYMMETRY. It uses a DB-25 input and has RCA outputs to go to a Punch or Power amplifier which amplifies SYMMETRY's signal.

RDAT The new illuminated RDAT is designed to be ergonomically easy to operate. It offers a non-volatile memory storage and is a transceiver to transmit and receive data from C.P.M.

By utilizing SYMMETRY today, Rockford is setting the pace for tomorrow's technology in mobile electronics.

THINKING ON YOUR FEET

You have chosen sales as your career and in doing so you accept the fact that your income is in direct correlation to your sales ability. Most professionals (doctors, lawyers, accountants etc.) spend many years training to receive the credentials and knowledge needed to foster a successful career. As a professional salesman, how much time have you spent learning the secrets of your trade? I refer to them as secrets because a close inspection of the audio salesman reveals that most of the basic sales skills are lacking. In fact, in most cases, sales people make critical mistakes that lead customers away from the sale. It may sound like we are preaching, but the simple fact is that these conditions represent a tremendous opportunity! What would happen if you chose to spend a small amount of your time sharpening your skills? You would join the small percentage of professional sales people who consistently outperform the pack. Carried one step further, if an entire organization were to expend some of its resources in this direction the effects could be astounding!

Now that the preaching is over, let's get to the meat of the subject. We have broken down your sales responsibility into four separate areas each of which are equally important. Just like math it is critical to learn each of the equation steps before proceeding to the next. Your introduction will lead smoothly into the qualifying ste which directs you to a proper presentation making the close a lot easier to swallow.

Notice how one step leads to another. Without an effective introduction you may never get the chance to qualify your customer and a good sales presentation, no matter how polished it may be, will fail if you are pitching something that won't fill the customers needs. **There is nothing more aggravating than listening to a sales pitch when you have no interest in the product that's being presented.** A big red flag immediately appears in the customer's mind that says **"Used car salesman."** If, however, you are sincerely interested in fulfilling a customers needs, not in selling him a used car, the odds are greatly increased when you go for the close. I'm sure it will will not be surprising to learn that there are some killer closing techniques that will help your customers part with their hard earned cash which is so vital to your pocketbook.

Steps To The Cash Register

I. Introduction

Q. Qualification

P. Presentation

T. Trial Close

C. Close

INTRODUCTION

Before you ever get a chance to say, "hello" to a customer they have already started thinking about you. What does your parking lot look like? What does your Yellow Pages ad look like? What does the glass on the front of your store look like?

Would it surprise you to know that the very first step in the sales process is where most critical mistakes are made? It is, believe it!

Proof of this is shown every time you go shopping for anything at a mall. The very first words to leave the salesperson's mouth 90% of the time is, "Can I help you?" Well as you know the used car salesman stereotype has helped program our response, and we almost always respond with, "No thanks, I'm just looking." SLAM! The door is closed on your potential sale, and the buyer just turned into a "Looker" who then, of course, becomes a "Be-back". Many customers in a buying mood can easily change their attitude if you avoid questions that invoke programmed responses, also referred to as "Yes & No" answers.

Never Say, "May I Help You!"

Since we are using a systematic approach to sales, let's list our objectives or purpose for using a proper introduction.

- The customer needs to feel comfortable in your store.
- The customer needs to feel comfortable with you.
- You need to establish some "common ground" or a rapport.
- You need insight into the customer's personality traits.
- You need to lead the customer into the qualification stage.

With these criteria in mind it is easy to see how important our introduction is to the entire process.

Here are some opening statement examples and questions that can point us in the right direction. Don't make your entire introduction a memorized, canned presentation. It needs to be sincere. Your customers will know. Always watch their eyes.

"Hi! How are you today?" "Have you been in our store lately?"

"Hi, I'm Joe. I'm just finishing up with another customer. Feel free to look around. If you have any questions, I will be available in a minute."

I think you get the idea. Basically, all you are trying to do is be perceptive and use our information gathering skills to help the customer feel comfortable. By developing that initial relationship we have set the stage for our qualifying process allowing us ask questions that would otherwise make the customer uncomfortable and feel out of place.

While carefully executing our introduction, we are also sizing up the customer by listening carefully. As we listen, we should also be perceptive to their body language and appearance. Who are we dealing with here? Is the person soft spoken and shy or loud and outgoing? Is this person highly professional or laid back and easygoing. All of these clues can be used to help the professional salesman relate with customers on their terms. Act professional with the professionals, relax with the surfers, and be crazy with the flamboyant .

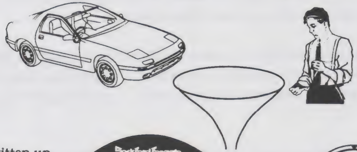
Now that we know how to make a proper introduction, let's move to the next step in our sales process.

QUALIFICATION

The qualifying stage of the sales process is basically an information gathering process. If done properly, when you are ready to give the actual presentation, you as a professional will know exactly which product will best fill the customers needs. If you start your presentation without a clear understanding of what the customer wants and needs, you will fumble around from one product to the next hoping to find a match. Most stores carry so many products that almost at once the customer is overwhelmed. *Here are some good points to remember when planning a sale:*

Your store carries all these products:

The customers' needs and the car act as a funnel:



What's left over is a sale ready to be written up.

Here are some steps that can help in the qualifying process.

- Find out what musical artists they like.
- Find out what equipment the customer is currently using.
- Find out what the customers like and dislike about their current equipment.
- Find out what improvements the customer would like to make.
- Find out what the customer's expectation for volume and sound quality are.

If you find yourself in a situation where the customers are not sure of what they want, then you can easily combine your qualifying stage with an educational process. The electronics industry is changing so rapidly that most consumers are not familiar with the latest bells and whistles or even the level of performance that is available.

When you find yourself in the above-mentioned situation, show the customers the top of the line. The newest, latest and greatest. Let the customers talk you down. Once you've said to the customers, "This will fit your needs," they can see no reason to spend more money.

Always Sell Down

High end car audio can present some difficult challenges in the qualifying process. As a salesman you must be able to gather all the information quickly and formulate creative solutions that will ultimately present an urgency for customers to buy.

There is a tried and true process that involves going out to the parking lot with the customers and looking at their cars. This is very effective because it gives you a chance to further qualify the customers and their specific needs. It also lets the customers know that you are genuinely concerned with meeting their needs. Who knows, you might find other topics of conversation that can help further your relationship. Always have a tape measure up front to take to the customer's vehicle. Using Term Pro on the sales floor is an incredible sales tool. It also establishes your store as the specialist.

If you determine that a subwoofer will be part of the sale, give a brief explanation of the different types of enclosures so you can determine how much space will be needed to satisfy the customers expectations of performance. From here you should take a quick look at the rest of the car to determine what options are available with regards to the rest of the system.

Occasionally you will find customers who have been given bogus advice by friends, or maybe have drawn their own conclusions, and will come in aski[illegible]r something that doesn't really meet their needs. In this case, the personal attention you give them [illegible] your rapport will enable you to educate them without offense.

Once you have determined the real need (large quantities of bass for example), you should ask a question like, "Are you open to a suggestion that I'm sure will improve your system?" Almost universally the answer is "Sure", and the customers are less likely to challenge your suggestion because they already stated that they are open to suggestions.

PRESENTATION

Product knowledge is the key to your success.

There is no worse feeling in retail than consistently not having the answers to customers' questions. Remember the last time you shopped for something and knew more about the product than the person who was presenting it? If they're a clerk it's justifiable. But if they are, as you are, a commission sales person you'd expect some product knowledge – at least more than yourself.

Read all you can about the products you sell. Use them to learn their abilities and inabilities. The more you know about your products the more you will sell. You'll have so much ammunition for your presentation you'll be able to easily paint a picture for your customers. Remember the funnel!

After you've gotten your products under your belt (not to be taken literally) you have to get a smooth presentation.

Demo Vehicles

The worst line in a presentation has to be, "This is exactly what I have in my car." If you can't or won't show your customers your system, then don't mention it. They won't care or believe you anyway.

Demo vehicles are very important. Whether it's your personal vehicle or a store owned demo car, you need them to make sales. Imagine going to a car lot and seeing 300 cars but not being allowed to test drive any of them. That's what your car stereo demo board looks like to your customers. Show them what a dedicated system sounds like.

TRIAL CLOSE

Testing The Water

Consumers may have a specific product in mind when they enter your establishment. For instance, "Do you have the new Punch 100DSM amplifier?" This question could be asked by a consumer that may have read a review from a mobile electronics magazine or they may have received a recommendation from a friend. Nevertheless, you should test the water by asking," Yes, we do. Would you like us to install it in your car today?" If this is unsuccessful, then start your introduction.

CLOSE

This is it! This is where so many very good sales presentations fail. You've done the job up to this point. You picked out the products that would fit the needs the customer said he needed filled. He's agreed with those selections. But then you have to ask for the money. You have to be in control. If he says, "Well let me think it over for nine or ten months...." You lost it. You should have found out the customers urgency in your qualification. Still you don't give up here and hand the customer your card. He'll be picking his teeth with it in less than five minutes.

Always Ask For The Money

Overcome your customers' objections. Maybe they aren't sure that their spouses will approve of the purchase. Insert you store's "Satisfaction Guaranteed" policy, whatever it may be. You might have more returns than other salespeople in your store, but you'll always have the highest closing ratio.

Something Extra At The Counter
Always give the customers that, "Jeez, I forgot to tell you about the best feature of all."

Make them feel great about parting with their money. If your store is having a promotion, save it until the end unless you need it as a close. For example: free install, giveaways, movie tickets, etc. Always point out something you missed (on purpose) during the presentation.

Follow up - Your best money maker

Turning Over The Keys
A professional salesperson's follow-up starts in the install bay. Always find the time to show your customers their install and new system. This also serves as prevention to customer abuse. Teach them about clipping, for instance. Arrange your customers' install when you're in. Not on your day off. Not when your shift is over. The more you make it clear that you want them to deal with you, the more likely that they'll let everyone else know they only deal with you, period.

Keeping Track
The last step is your own file box. Keep track of your customers' systems and what they need next. Their preference in music. Even their personal statistics (ie. wife & kids names). Keep them bringing in referrals to you, especially during events. When your store is busy it creates an atmosphere of buying. Things are happening. It is exciting to be in the store during special events and it's great to have return customers selling to their friends for you.

If you choose to only work in the audio business as a part timer or as a hobbyist satisfying an addiction, you owe it to yourself to be the best you can be. You also owe your store owner your most sincere effort you can put forth. Besides, you will have had a great experience. Think of all the knowledge you will have gained. And remember to "think on your feet."

GLOSSARY

Attenuator
This is used for increasing or decreasing the strength of a signal.

Bandwidth
Refers to the "space" in the frequency response of a device through which audio can pass (800Hz-3kHz)

Basket
The foundation by which most of the other parts are supported. Its most important function is to provide proper strength and support without adding unwanted resonances that could potentially color the sound quality. The most common types are stamped out of metal or cast out of aluminum or other alloys.

Bi-Amplification
A "bi-amped" system utilizes multiple amplifiers for woofers, midrange and tweeters in a speaker system.

Bi-Polar
An "N' device with a small "P' device in the center that uses current to make the component work. As the current builds up on the "P" device, it allows the signal to pass through. Bipolar devices require more negative feedback which degrades the sound. These devices are prone to thermal runaway requiring very complex protection circuitry which also degrades the sound.

BL Product
Measured in Tesla meters. It is the magnetic flux density parameter and is the primary energy factor in a dipole transducer or moving coil loudspeaker.

Bottom and Top Gaskets
These gaskets are needed for proper installation of the speaker. They are made of foam and chip board. Eliminating one or the other gasket is strictly a cost savings by the manufacturer.

Class A Amplifiers
This is when the output current conducts 100% of the time, thereby allowing current to flow through both devices during an entire cycle of signal.

Class B Amplifiers
Each device draws current half of each cycle while the other device is cut off.

Class A/B Amplifiers
Current flows through each device during more than half of each cycle, but cuts off during the rest of the cycle while the other device takes over.

Cone
The cone is generally made of paper or any number of different types of plastics. Its function is to move in and out like a piston creating the acoustical sound waves. The cone must be rigid enough to handle power yet not be so heavy that it restricts the movement of the former in the magnetic gap. At the same time a cone that is not rigid will buckle under high power situations thus adding distortions of its own.

Crossover Point
The frequency at which the crossover has reduced it 3dB (only Butterworth and Bessel alignments).

dB
The decibel is a unit of measurement for ratios of sound level, power, voltage, and other quantities.

Digital Time Delay
This electronically delays the audio signals in milliseconds resulting in an effect much like one experiences in a concert hall.

Dust Cap
Glued to the cone, the dust cap keeps unwanted dust and debris out of the magnetic gap and away from the voice coil. A porous dust cap allows air to flow through the magnetic field to cool the voice coil.

Frequency
The change in current or voltage in an electrical signal or of air pressure in an acoustical signal (sound).

F3
Where the response of the speaker is 3dB down from its cutoff point.

FS
The free air resonance of a driver is expressed in Hertz. It is the frequency in which the driver resonates naturally.

Hertz
The unit for measuring cycles per second.

kHz
Abbreviation for kilohertz, or one thousand cycles per second.

Loosely Regulated
Allows the amplifier to deliver more current and voltage as input voltage and current is increased.

Magnet
The magnets primary function is to provide a constant magnetic field around the voice coil. It also serves as a heat sink to dissipate unwanted heat that could damage the voice coil. Magnets are made from different mined ores. Subsequently, they are different strengths.

MMs
The mass of a moving system that consists of a coil, cone, one half the surround and the air load.

MOSFET - Metal Oxide Field Effect Transistor
MOSFET uses both current and voltage gain in the amplification process. Each MOSFET is either a "P" type or an "N" type device. Voltage builds up on the conductive layers being filtered by the silicon glass insulator. Voltage is allowed through and equal amounts of current can pass through the previously inert material. MOSFETs don't require large amounts of negative feedback because the circuitry is much less complicated using fewer parts.

Octave
A unit of measure on a scale labeled in Hertz. An octave higher than 400Hz is 800Hz and an octave lower is 200Hz.

PE
The driver's continuous power rating in watts.

Pole Piece
The pole piece is connected to the bottom plate and is slightly smaller than the circular hole in the magnet, creating the magnetic gap in which the voice coil former slides into.

Q (The Quality Factor)
A factor that determines the characteristic of a system's resonance.

QES
The driver's electrical "Q."

QMS
The driver's mechanical "Q."

QTS
Describes the combination of QES and QMS. The driver's "Q" at resonance.

RE
The DC resistance of the driver's voice coil.

REF
Sensitivity is measured as dB SPL at 1 watt of power across the speaker's nominal voice coil impedance at a distance of 1 meter.

Spider
The spider is a part of the suspension whose primary function is to keep the voice coil centered in the gap between the magnet and pole piece. It also brings the speaker back to its center position when no electrical current is flowing through the voice coil.

SD
The effective surface area of the cone.

Surround
Glued to the cone and basket the surround serves as part of the suspension of the speaker. It is usually made of compressed foam or rubber. This also aids in controlling the cone in over excursion situations.

Tightly Regulated
Regulates the amount of voltage and current making sure that the specified rated output is always delivered.

Top Plate

The top plate is attached to the top of the magnet to hold the basket and magnet structure together.

VAS

Expressed in cubic feet or liters. It is the volume of air required for the driver to achieve its FS.

Voice Coil

This is a coil of insulated wire that is wrapped around a former and rests centered in the middle of the magnet assembly. It is glued to the cone so that its movements are transferred accurately from the coil to the cone. Voice coil wire varies from round to partially flat and comes in many different gauges.

XMAX

Measured in millimeters. The maximum distance a voice coil can move in one direction before the coil moves away from the magnetic field.

Z

The resistance of a loudspeaker's voice coil.

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UNDERSTANDING THE ENVIRONMENT

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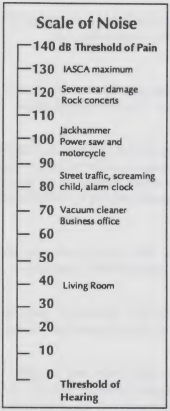
- Symbols & Orders of Magnitude

Understanding the Environment

The next time you go to a dance, rock concert, or night club where the music will probably be loud, try this interesting test of your hearing. Before leaving your car, turn off the engine, roll up the windows, and play a tape or CD over your car audio system. Set the volume control to where the music is *just barely audible.*

When you get back from the evenings exposure to loud music, before you start your car engine or roll down the windows, play the same selection of music without changing the volume setting. If the event was typical, you probably won't be able to hear the music until you raise the volume from its previous setting.



This partial deafness is called "temporary threshold shift" (TTS), a reduction in your hearing acuity. While usually temporary, threshold shift can be permanent if your ears are sufficiently abused.

You already know that loud sounds can damage your hearing: using a jackhammer or working near a generator does little to help our hearing acuity. But too few music lovers realize even normal, day-to-day activities can also cause hearing damage. In addition to mowing the lawn or using other power tools, these activities include the musical ones mentioned in the above experiment.

During recent years of increasingly powerful electronic amplifiers and loudspeakers, it appears more professional musicians, recording engineers and audiophiles are suffering from temporary or permanent hearing loss. Rock stars such as Peter Townsend, Rod Stewart, and Ted Nugent have gone partially deaf or have complained about loss of hearing. There are *many* other rock stars suffering from a condition known as tinnitus (ringing in the ears) from loud music. But, rock music is not at issue alone. Any music, even classical music, can be too loud. Recent studies have discovered many orchestral musicians have also suffered sound-induced hearing losses.

Unfortunately, human ears never "get used to loud sounds," you just go deaf, so the argument that you get used to loud sounds really doesn't make much sense. The fact of the matter is, your ears are the best indicator of what's too loud. And, "too loud" means specifically sounds loud enough to cause permanent damage if endured for any length of time.

Here are three warning signs:

1. If, after exposure to loud noises or music, your high-frequency hearing is temporarily impaired so ordinary sounds seem muffled, then those noises were too loud.
2. The sounds were too loud if you hear ringing in your ears after exposure.
3. If you have to shout to a person standing an arm's length away or have to yell directly into someone's ear in order to make yourself heard, the ambient sound is too loud.

If further exposure to excessively loud sounds continues, both the temporary muffling of high frequencies and ringing of the ears can develop into permanent deafness.

Decibels, Hertz & Sound Pressure Levels

As you are probably aware, sound is generated by the movement of air set in motion by vibrating objects. The strength, or loudness, of these vibrations is measured in decibels, signified by "dB", based on a logarithmic scale similar to the Richter scale used in measuring the intensity of earthquakes. For instance, the sound of a vacuum cleaner (70dB) produces 10 times more acoustic energy than the sound of two people talking in a room (60dB). Sounds become painfully loud somewhere between 125dB and 140dB.

Hertz (Hz) are the method of measuring the length of a wave cycle, with one wave cycle representing one Hertz. Low sounds have longer wave cycles, and therefore a smaller Hz, while high sounds have much shorter wave cycles, and higher Hz.

Sound Pressure Level (SPL), is another method of measuring sound levels in the air. You frequently find this measurement working together with decibels. While decibels measure the loudness of sound, SPL measures the relative pressure on our eardrums caused by sound. The reason we use both decibels and SPL to measure sound is that these methods of measurement best represent the way we hear sound levels. An increase of 10dB SPL is perceived as being the same relative change in loudness whether it is between 40 and 50dB SPL or between 100 and 110dB SPL.

Protect Yourself

Frequency content, duration of exposure, and *level* are the three factors which determine the danger of any sound, so altering any of them can protect your hearing from temporary or permanent damage.

Frequency Content

If you could somehow change the frequency content of a sound without changing its musical or other values, you could significantly reduce its potential for causing hearing damage. This is because the ear is more vulnerable to certain frequency ranges than others.

To take an extreme example, you could use an equalizer to effectively remove all the frequencies around 4,000Hz which is the frequency range of the ear most easily damaged. Unfortunately, while this technique is used in the application of sound dampening in industrial environments, it is a poor technique for music, since many of the desirable frequencies in voice and instrument performance are around the 4,000Hz range.

Duration Of Exposure

Usually you can limit the amount of time you are exposed to very loud sounds. This is a very effective way to protect your hearing, since it gives the hearing mechanism a chance to recover from the stresses it undergoes during exposure to loud sounds.

Federal government (OSHA) guidelines state that with every 5dB increase in SPL, you must *halve* the duration of sound exposure starting from 8 hours at 90dB (this means a maximum of 4 hours at 95dB, 2 hours at 100dB), and down to 15 minutes or less at 115dB.

Although these guidelines are meant to protect employees at the workplace, they apply just as easily to critical music listening. In fact, since they are intended to protect against permanent hearing damage, they are probably too liberal to protect the listener from the temporary threshold shifts which may affect critical sonic judgements.

Car audio enthusiasts should remember the "15-minute/115dB guideline." This can easily be exceeded during a sound-off when measured levels from some systems can top out at over 120 dB. You shouldn't listen to more than a few minutes at such levels. Many sound-off judges have adopted an "earplug" policy for their own ear protection.

Overall Sound Level

A reduction in the overall sound level is the most practical and effective way to protect your hearing. Simply lowering the volume control by 5dB will allow you to spend twice as much time doing critical listening. Putting distance from the source of sound also helps: every doubling of distance reduces your exposure by around 6dB, again increasing the allowable exposure time. This rule works best when you are relatively close to the sound compared to the size of the listening environment.

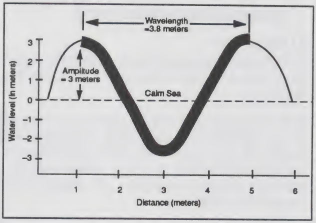
One of the best methods for lowering risk is by using earplugs which reduces sound levels of 120dB down to 100dB. You can spend two hours enjoying the music without having to get up to give your ears a rest.

WHAT ARE SOUND WAVES?

Our world is made up of many kinds of waves. You've often heard the terms "radio waves," "microwaves," "sound waves," or other kinds of waves. Imagine for a moment sitting on the beach, watching the waves break on shore. The waves are constant, and have a rhythm all their own. You hear the sound created by the roaring water as the waves break. If you're close enough to the breakers, you might even feel the vibrations caused by the waves. When the waves are large and powerful, the sound might remind you of rolling thunder. When the waves are small and much less powerful, the sound they make is different, and not as loud. There is a strong relationship between sound and vibration, as we will discover.

Sound travels in much the same manner as waves in the ocean. However, there are many differences between these two kind of waves. Among these differences is speed. While water waves (currents) are measured in miles per hour, kilometers per hour, or knots, sound waves travel at 344 meters per second! But, compare the speed of sound waves with the speed of light which travels at an incredible 300,000 kilometers per second! One important key to understanding waves is found in the power of the wave. The power associated with the wave is referred to as the *relative power output* of the wave, or the force of the wave.

Another significant aspect of both sound waves and waves in the ocean can be characterized by the term *amplitude.* Amplitude is measured as the distance from the flat position (a calm sea) to the crest of the wave. In addition, it is also a measurement of the distance from the flat position to the bottom of the trough. As you can see, the power of a wave with a measurement of 12 feet is much more powerful than a 2-foot wave. It is also safe to conclude that a large wave carries much more energy than a small wave.



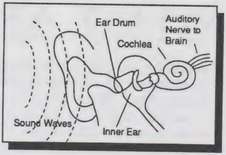
Wavelength is another characteristic which we should explore. Wavelength is defined as the horizontal distance between two successive crests in waves. It may also be expressed as the horizontal distance a wave travels during a single cycle (may be measured at any identical point of a wave, but usually at the crest, flat position, or bottom of the wave).

When we combine these three elements of waves we have created the same set of conditions that exist in musical waveforms: a relative power output, an amplitude of the waveform, and a specific wavelength.

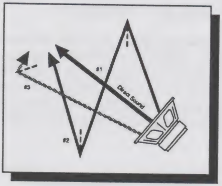
Making Sound

Sound is made by making something vibrate. As a vibrating object moves, it sets up sound waves in the air. The waves consist of alternating regions of high and low pressure, which are called *compression* and *rarefaction.* As the object's surface moves forward into the air, it produces a compression. The surface then moves back, producing a rarefaction. Together each compression and rarefaction makes up a sound wave, and the waves move out in all directions at high speed (344 meters per second). The stronger the vibrations, the greater the pressure difference between each compression and rarefaction and the louder the sound.

Hearing



The vibration needed to create an audible sound wave has to have a rate of more than 20 compressions and 20 rarefactions per second. As sound waves enter the ear, the pressure differences between successive compressions and rarefactions set the ear drum vibrating. These vibrations pass to the cochlea in the inner ear, where they are converted into electric signals. The signals travel along the auditory nerve to the brain, and the sound is interpreted and heard.



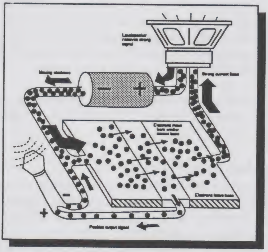
Sound Reflections

Sound waves may bounce off nearby surfaces. The ear receives a mixture of the direct sound and echoes. If the reflecting surfaces are fairly distant, the reflected sound will take much longer to reach the ear and separate echoes will be heard.

Electricity

All things are made up of atoms, and within atoms are smaller particles called electrons. Electrons each have an electrical charge and this charge, which is considered to be negative, is the fundamental cause of electricity.

Static electricity is so-called because it involves electrons which are moved from one place to another rather than ones which flow in a current. In an object with no static electric charge, all the atoms have their normal number of electrons. If some of the electrons are then transferred to another object by, for example, vigorous rubbing or brushing, the other object becomes negatively charged while the object that loses electrons becomes positively charged. An electrical field is set up around each object.



"Unlike" charges always attract each other and "like" charges always repel each other. Rubbing or brushing creates a charge and therefore an electric field. The field affects objects nearby, producing an unlike charge in them, and the unlike charges are drawn together.

Current electricity is produced by electrons on the move. Unlike static electricity, current electricity can only exist in a conductor, a material such as a metal, that allows electrons to pass freely through it.

In order to make electrons move, a source of energy is needed. This energy can be in the form of light, heat, or pressure, or it can be the energy produced by a chemical reaction. Chemical energy is the source of power in a battery-powered circuit.

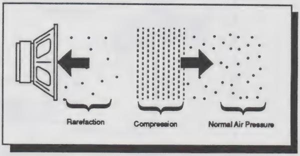
Electronic Amplification of Music

Another technique to enhance sound quality with increasing loudness is through electric amplification. An ideal example of sound amplification by electronic means is the electric guitar. Without electronic amplification the electric guitar body does not have the resonance of an acoustical guitar body, and is much less audible. When electronic amplification is supplied, playing the metal guitar strings causes pick-ups beneath the strings to generate electrical sound signals. The signals go to volume and tone controls, which determine the loudness and kind of sound, then to an amplifier and loudspeaker system.

Electronic Mixer
An electronic mixer takes sound signals from several different sources and mixes them together. The tone and volume of each signal is controlled so that a good sound balance results. One combined (or two for stereo sound) signal, then goes to the amplifier and loudspeakers.

Amplifier
An amplifier increases the voltage of a weak signal from a microphone, mixer, electric instrument, radio tuner or tape or disc player making it powerful enough to drive a loudspeaker or earphone. It works by using the weak signal to regulate the flow of a much stronger current which normally comes from a battery or the electricity supplied. The key components that regulate the flow of the strong current are usually transistors. An amplifier usually contains many transistors and other components that enable the amount of amplification and also the tone of the sound to be varied.

Rarefaction in Sound Waves
The area in between each wave of sound is a trough of less-dense air. This is called rarefaction. It is also the area of sound waves where the sound pressure level is smallest.



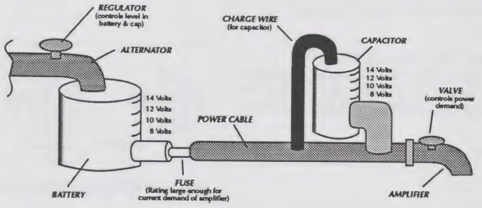
Compression in Sound Waves
When the microphone diaphragm is pushed in by a compression in the sound wave , it reverses the flow of electrons in the weak signal. Electrons leave the base semiconductor in the center of the sandwich and create holes. Forced by the power supply, many electrons enter these holes from the emitter and then move on into the collector. The result is a flow of electrons much larger than that in the weak signal, but exactly in step with it: the weak signal has been amplified.

Loudspeaker
A loudspeaker reproduces sound by responding to the electrical signal produced by the amplifier. Most loudspeakers are "dynamic." They contain a thin but rigid cone which is vibrated by the movement of a coil. The signals from the amplifier are fed to the coil, which sits inside a magnetic field created by a circular permanent magnet. The coil produces its own magnetic field as the signal current passes, and the fields create movement (vibration) which produces sound.

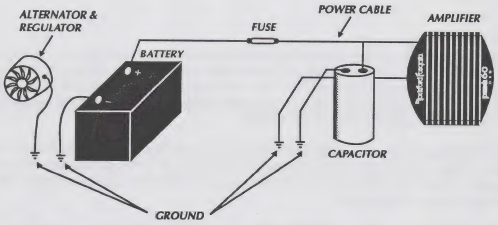
An electrostatic loudspeaker works in a different way. It contains a large vibrating diaphragm which is given a strong electric charge. The signal from the amplifier is stepped up in voltage by a transformer and applied to two perforated plates on either side of the diaphragm. The resulting electrostatic field makes the diaphragm vibrate and produce sound.

ELECTRICAL SYSTEM

Voltage	=	Height of water in the battery and capacitor / pressure in lines
Capacitance	=	Volume of water stored in the battery and capacitor
Current	=	Rate of flow of the water (through the water pipes)
Power	=	Amount of water passed through the system (watts)
Resistance	=	Size of water lines, any reduction in the flow of water (ohms)

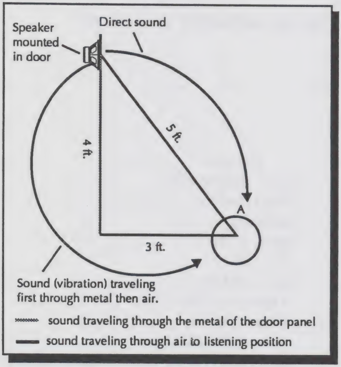


WATER ANALOGY



UNDERSTANDING THE ENVIRONMENT

One way to maximize your environment is to provide a good platform for your speaker mounting. When mounting the speaker on door panels, not only is it important to have a secure mounting location, but it's also important to prevent the vibration of the speaker from transmitting down the door panel. What effect could this vibration cause? Below are the arrival times of direct sound of the speaker and sound (vibration) transmitted first down the metal panel.



Question: Which sound will arrive first?
Givens: Speed of sound in air (STP) - 1130 ft./sec.
Speed of sound in metal - 16,600 ft./sec.

Medium of Sound	Speed of Sound Ft/Sec	Meters/sec.
Air (STP)	1,130	344
Sea Water	4,900	1,500
Wood (fir)	12,500	3,800
Steel (bar)	16,600	5,050
Gypsum Board	22,300	6,800

$$\boxed{\text{Direct Sound}} \;-\; \frac{5\,\cancel{ft.}}{1130\,\cancel{ft}/sec} = 0.0044 \text{ or } \boxed{4.4\text{ ms}}$$

$$\boxed{\text{Door Panel}} \;-\; \frac{3\,\cancel{ft.}}{1130\,\cancel{ft.}/sec} + \frac{4\,\cancel{ft.}}{16{,}600\,\cancel{ft.}/sec.} = .00290\text{ sec. or } \boxed{2.9\text{ms}}$$

The sound traveling through the metal of the door panel reaches the listening position ahead of the radiated "direct" sound of the speaker. If an analogy of two speakers is used, the difference in time arrival (1.5ms) would equal a difference in distance of 1.695 feet between the two speakers relative to the listening position. Obviously, the resonance of the door panel will add an inverse affect in sound quality. By damping the panel (with Punch mat or other damping material) the effect of this door resonance can be reduced at least 60dB* below the music level. Damping panel resonances will provide the platform necessary for awesome stereo reproduction.

* 60dB is equal to the human threshold of hearing. If a test tone A is $\geqq$ 60dB than test tone B, then test tone A will be loud enough that it will mask or "wash out" test tone B. Mini-disk utilizes this knowledge of human hearing to compress the digital audio signal. By eliminating all musical information that is 60dB below other musical information, the saved space on the disk enables this format to have the same playing time as compact disc but be considerably smaller in size.

SYMBOLS & ORDERS OF MAGNITUDE

Symbols

F = Farads:	measurement value for capacitors
R, Ω = Ohms:	measurement value for resistors
H = henries:	measurement value for inductors
f = frequency:	measurement of sound representing cycles per second of air movement
I, A = amperes:	measurement value for current
E, V = voltage:	measurement value for voltage

Orders of Magnitude

Term	Symbol	Magnitude	Example
mega	M	10^6	6MΩ = 6,000,000Ω
kilo	k	10^3	7km = 7,000km
hecto	h	10^2	4hl = 400l
deca	da	10^1	2dam = 20m
deci	d	10^{-1}	8dl = 0.8l
centi	c	10^{-2}	42cm = 0.42m
milli	m	10^{-3}	6.6mH = 0.0066H
micro	μ	10^{-6}	100μF = 0.0001F
nano	n	10^{-9}	27nm = 0.000000027m
pico	p	10^{-12}	9pf = 0.000000000009f

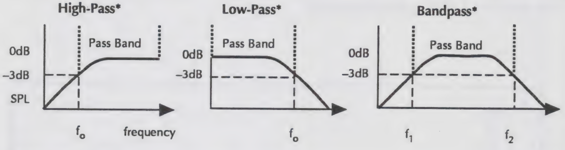
PASSIVE COMPONENT SYSTEMS

- Understanding Components
- Crossover Slopes & Types
- Frequency / Impedance Shaping Networks

WHAT ARE CROSSOVERS

A network filter is a device that modifies an audio signal as a function of frequency. Crossovers are one form of network filter. Filters divide an audio signal and send those bandwidths of signal to dedicated speakers / amplifiers.

Three Types Of Crossovers



*Q of Crossover = .707

In a high-pass crossover, high frequency information is allowed to pass and low frequency information is filtered off as frequency decreases. A low-pass crossover performs the inverse of a high-pass crossover. At low frequencies, a low-pass crossover (sometimes referred to as filter) passes information and attenuates the output as frequency increases. A bandpass crossover filter is a combination of a high-pass and a low-pass filter. Crossovers can be Digital, Active and Passive.

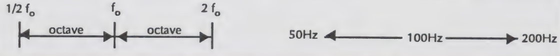
Crossover Slopes

Crossover slopes are referred to as dB/octave which is the difference in the output level (dB) versus the change in octaves (frequency).

Definitions:

dB Measure of loudness. +10dB louder is perceived as twice as loud; +3dB is an audible difference. It takes twice the power to achieve 3dB acoustic gain and ten times the power to achieve a 10dB acoustic gain.

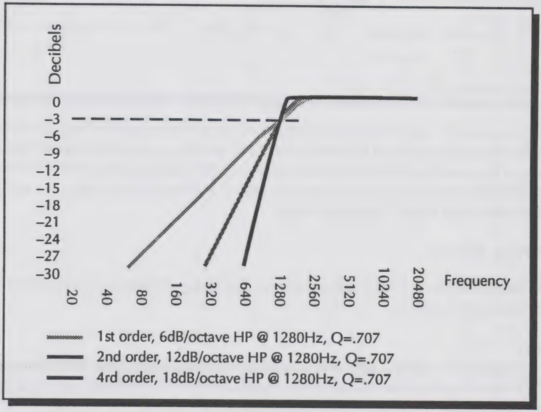
Octave The distance from a certain frequency defined as the halving or doubling of the starting frequency.



For example: 50Hz is 2 octaves away from 200Hz.

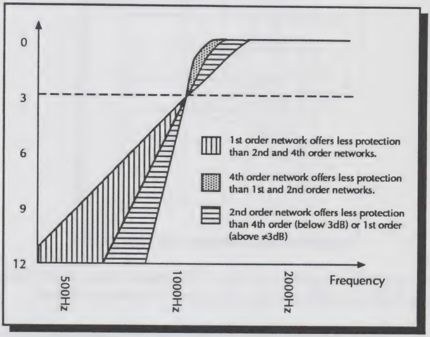
Crossover order = (n) 6dB/octave

ex. 1st order = 1 x 6dB/octave slope = 6dB
2nd order = 2 x 6dB/octave slope = 12dB
3rd order = 3 x 6dB/octave slope = 18dB
4th order = 4 x 6dB/octave slope = 24dB

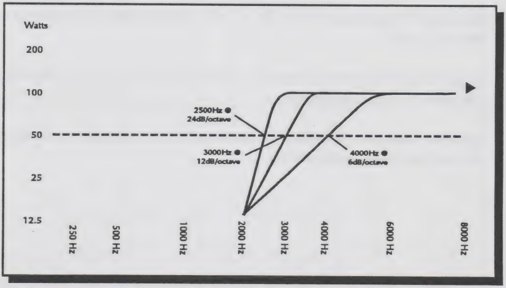


The advantage of using higher order slopes is extended bandwidth of driver (speaker) operation while maintaining driver protection.

Below is an application of the benefits of higher order crossovers.



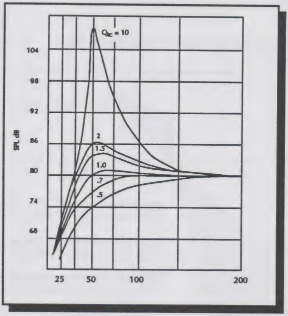
The tweeter can handle only 12.5 Watts @ 2kHz but 100 Watts @ 3.0kHz.



All three crossover points offer the same protection at 2000Hz.

Q Response

The Q Response refers to the "shape" of the crossover roll-off at the crossover frequency (Figure 1).

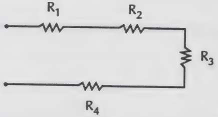


Different Q Responses affects the phase response of the speaker system as the high-pass and low-pass responses are summed together. Different alignments have different applications.

Butterworth (Q = .707)	Maximum flat response best suited for speakers mounted on the same plane on axis with the listening position.
Bessel (Q = .58)	Best suited for speakers mounted 90° of axis from each other, i.e., midrange in door, tweeter in dash.
Linkwitz-Riley (Q = .49)	Response optimizes speakers mounted on the same plane relative to each other, but off axis from the listening position.
Chebychev (Q = 1.0)	Response designed to compensate for driver performance irregularities inherent in less than optimal mounting locations.

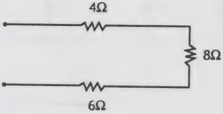
Series Circuits

$$R_T = R_1 + R_2 + R_3 \ldots$$



Example:

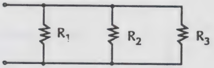
$$R_T = 4\Omega + 8\Omega + 6\Omega = 18\Omega$$



Note: R_T is always larger than the largest resistance in the circuit, i.e. $R_T = 18\Omega > 8\Omega$

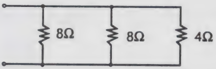
Parallel Circuits

$$R_T = \frac{1}{\frac{1}{R_1} + \frac{1}{R_2} + \frac{1}{R_3} \ldots}$$



Example:

$$R_T = \frac{1}{\frac{1}{8} + \frac{1}{8} + \frac{1}{4}} \Rightarrow \frac{1}{\frac{2}{8} + \frac{1}{4}} \Rightarrow$$



$$\frac{1}{\frac{1}{4} + \frac{1}{4}} \Rightarrow \frac{1}{\frac{2}{4}} \Rightarrow 2\Omega$$

Note: R_T is always smaller than the smallest resistance in the circuit, i.e. $R_T = 2\Omega < 4\Omega$

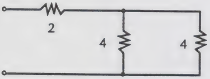
For two resistors:

$$R_T = \frac{R_1 \times R_2}{R_1 + R_2}$$

Series & Parallel Circuit Combinations

Start from the furthest point relative to the measurement point and combine groups of resistors (series and parallel connectors).

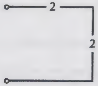
Example:



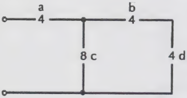
Step 1:

$$R_T = \frac{R_1 \times R_2}{R_1 + R_2} = \frac{4 \times 4}{4 + 4} = \frac{16}{8} = 2\Omega$$

Step 2:



$R_T = R_1 + R_2$ or $2 + 2 = 4\Omega$



Step 1: Combine resistors b & d.

$4 + 4 = 8$

Step 2: Combine c with the answer to Step 1.

$$\frac{8 \times 8}{8 + 8} = \frac{64}{16} = 4\Omega$$

Step 3: Combine a with answer to Step 2.

$4 + 4 = 8\Omega$

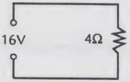
$V = IR$

Voltage = Current x Resistance

$R = \frac{V}{I}$ and $I = \frac{V}{R}$

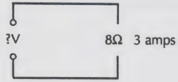
Note: These laws apply to both AC (alternating current) and DC (direct current) circuits.

Example:



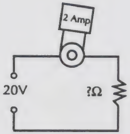
$V_T = 16V$
$R_T = 4\Omega$
$I_T = ?$

$I = \frac{V}{R}$ or $\frac{16}{4} = 4$ Amps



$V_T = ?$
$R_T = 8\Omega$
$I_T = 3$ amps

$V = IR$
$= (8)(3)$
$= 24$ Volts



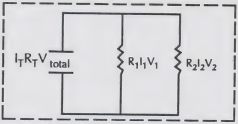
$V_T = 20$
$R_T = ?$
$I_T = 2$ amps

$R = \frac{V}{I} = \frac{20}{2} = 10\Omega$

Things to remember for parallel circuits.

1. Voltage remains constant.

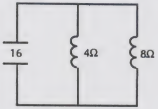
   $V_{total} = V_1 = V_2 ...$



2. Total Current = Sum of individual currents in system.
   $I_{total} = I_1 + I_2 + ...$

3. The lower the impedance, the greater the current demand from the system.

4. Of two resistive loads, the load with the lower impedance will receive more current (#3) and therefore more power (Watts). Power is a function of resistance and current.



1. Find current of each load. 4Ω (a) and 8Ω (b)
2. Find total current demand in system.
3. Find power for each load. 4Ω (a) and 8Ω (b)
4. Find total power in system.

1. $V = IR \Rightarrow I = \frac{V}{R}$

   Load a $I = \frac{16}{4} = 4$ amps Load b $I = \frac{16}{8} = 2$ amps

2. $I_{total} = Ia + Ib = 4 + 2 = 6$ amps

3. Power = $\frac{V^2}{R}$ or I^2R

   Load a $\frac{16^2}{4} = \frac{256}{4} = 64$ Watts or $4^2 \bullet 4 = 16 \bullet 4 = 64$ Watts

   Load b $\frac{16^2}{8} = \frac{256}{8} = 32$ Watts or $2^2 \bullet 8 = 4 \bullet 8 = 64$ Watts

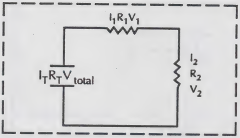
4. $Power_{total} = Pa + Pb... \Rightarrow 64 + 32 = 96$ Watts
   or
   $Power_{total} = \frac{V^2_{total}}{R_{total}}$ $R_{total} = \frac{1}{\frac{1}{4} + \frac{1}{8}} = \frac{8}{3}$

   $\frac{16^2}{\frac{8}{3}} = \frac{3 \bullet 16^2}{8} = \frac{3 \bullet 256}{8} = 3 \bullet 32 = 96$ Watts

Things to remember for series circuits.

1. Total Voltage = Sum of individual voltages in system.

   $V_{total} = V_1 + V_2 + V_3 \ldots$

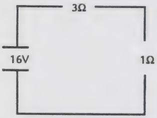


2. Current remains constant.
   $I_{total} = I_1 = I_2 = I_3 \ \ldots$

3. Resistance total = sum of individual resistors
   $R_{total} = R_1 + R_2 + R_3 \ldots$

4. Power in a series circuit. Since current remains constant…
   $P = I^2 R$

5. Of two resistive loads, the higher impedance load will get more power.



1. Find total current.
2. Find voltage at each load. 3Ω (a) and 1Ω (b)
3. Find power for each load.
4. Find total power in system.

1. $V_t = 16$ $\quad I_t = \dfrac{16}{4} = 4$ Amps
   $R_t = 4\Omega$

2. Load (a) $V = IR = 4 \bullet 3 = 12V$

   Load (b) $V = 4 \bullet 1 = 4V$

3. Load (a) $P = I^2R$; $P = 4^2 \bullet 3 = 48W$

   Load (b) $P = 4^2 \bullet 1 = 16W$

4. Total Power
   $P_t = Pa + Pb \Rightarrow 16 + 48 = 64W$
   or
   $P_t = \dfrac{V^2}{R} \Rightarrow \dfrac{16^2}{Ra + Rb} = \dfrac{16^2}{3 + 1} =$

   $\dfrac{16^2}{4} = 64W$

FORMULAS
FORMULAS
FORMULAS

OHM'S LAWS

For Parallel Circuits	For Series Circuits
$R_t = \dfrac{1}{\frac{1}{R_1} + \frac{1}{R_2} + \frac{1}{R_3} \cdots}$ $R_t = \dfrac{R_1 \times R_2}{R_1 + R_2}$ (for 2 resistors)	$R_t = R_1 + R_2 + R_3 \cdots$

VOLTAGE LAWS

$V = IR$ (Voltage = Current x Resistance)

$V = IR \qquad \dfrac{V}{R} = I \qquad \dfrac{V}{I} = R$

POWER LAWS

$P = VI$ (Power = Voltage x Current)

$P = VI \qquad P = \dfrac{V^2}{R} \qquad P = I^2R$

REACTANCE LAWS

For inductors

$X_L = 2\pi fL$

For capacitors

$X_C = \dfrac{1}{2\pi fC}$

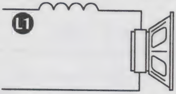
where: L is the inductance in Henries
C is the capacitance in farads
f is the frequency at the testing point

dB LAWS

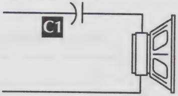
$\Delta dB = 20 \log \left(\dfrac{V_2}{V_1}\right) \qquad \Delta dB = 10 \log \left(\dfrac{P_2}{P_1}\right)$

$\Delta dB = 10 \log \left(\dfrac{V_2{}^2}{V_1{}^2}\right) \qquad \Delta dB = 20 \log \left(\dfrac{R_1}{R_1 + R_2}\right)$ (for series circuit)

6dB / Octave High and Low-Pass Filters



6dB/Octave Low-Pass



6dB/Octave High-Pass

Formulas

$$L = \frac{R}{2\pi f}$$

$$C = \frac{1}{2\pi f R}$$

L = inductor in Henries
C = capacitor in farads
f = frequency
R = resistance of speaker

Table of Component Values

Frequency Hertz	Speaker Impedance					
	2 Ohms		4 Ohms		8 Ohms	
	L	C	L	C	L	C
80	4.1mH	100µF	8.2mH	500µF	16mH	250µF
100	3.1mH	800µF	6.2mH	400µF	12mH	200µF
130	2.4mH	600µF	4.7mH	300µF	10mH	150µF
200	1.6mH	400µF	3.3mH	200µF	6.8mH	100µF
260	1.2mH	300µF	2.4mH	150µF	4.7mH	75µF
400	.8mH	200µF	1.6mH	100µF	3.3mH	50µF
600	.5mH	136µF	1.0mH	68µF	2.0mH	33µF
800	.41mH	100µF	.82mH	50µF	1.6mH	26µF
1000	.31mH	78µF	.62mH	39µF	1.2mH	20µF
1200	.25mH	66µF	.51mH	33µF	1.0mH	16µF
1800	.16mH	44µF	.33mH	22µF	.68mH	10µF
4000	.08mH	20µF	.16mH	10µF	.33mH	5µF
6000	51mH	14µF	.10mH	6.8µF	.20mH	3.3µF
9000	34mH	9.5µF	68mH	4.7µF	.15mH	2.2µF
12000	25mH	6.6µF	51mH	3.3µF	100mH	1.6µF

Note: These values are approximate for pure resistive loads only. Speakers vary in impedance with frequency and type, and will not match these values exactly.

BUTTERWORTH ALIGNMENT Q = .707

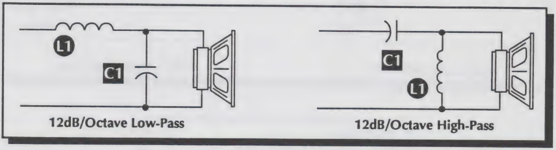


Table of Component Values

Frequency Hertz	Speaker Impedance					
	2 Ohms		4 Ohms		8 Ohms	
	L	C	L	C	L	C
80	5.6mH	700µF	11mH	330µF	22mH	180µF
100	4.5mH	500µF	9.1mH	270µF	18mH	150µF
130	3.5mH	470µF	6.8mH	200µF	15mH	100µF
200	2.3mH	330µF	4.7mH	150µF	9.1mH	75µF
260	1.7mH	220µF	3.6mH	100µF	6.8mH	50µF
400	1.1mH	140µF	2.2mH	68µF	4.7mH	33µF
600	.75mH	100µF	1.5mH	47µF	3.0mH	26µF
800	.56mH	68µF	1.0mH	33µF	2.0mH	15µF
1000	.45mH	55µF	.91mH	27µF	1.8mH	13µF
1200	.38mH	47µF	.75mH	22µF	1.5mH	11µF
1800	.25mH	33µF	.50mH	15µF	1.0mH	6.8µF
4000	.11mH	14µF	.22mH	6.8µF	.47mH	3.3µF
6000	75mH	10µF	.15mH	4.7µF	.33mH	2.2µF
9000	50mH	6µF	.10mH	3.3µF	.20mH	1.5µF
12000	38mH	4.7µF	75mH	2.2µF	.15mH	1.0µF

Note: These values are approximate for pure resistive loads only. Speakers vary in impedance with frequency and type, and will not match these values exactly.

18dB / Octave High-Pass Filters

BUTTERWORTH ALIGNMENT Q = .707

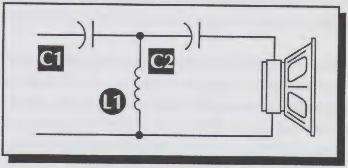


Table of Component Values

Frequency Hertz	Speaker Impedance 4 Ohms C1	4 Ohms L	4 Ohms C2	8 Ohms C1	8 Ohms L	8 Ohms C2
80	330µF	6.0mH	1000µF	160µF	12mH	500µF
100	270µF	4.7mH	800µF	150µF	10mH	400µF
130	200µF	3.9mH	600µF	100µF	7.5mH	300µF
200	130µF	2.4mH	400µF	68µF	5.4mH	200µF
260	100µF	1.8mH	300µF	50µF	3.3mH	150µF
400	68µF	1.2mH	200µF	33µF	2.4mH	100µF
600	47µF	.80mH	130µF	91µF	1.6mH	68µF
800	33µF	.60mH	100µF	16µF	1.2mH	50µF
1000	27µF	.47mH	75µF	13µF	.90mH	39µF
1200	22µF	.39mH	68µF	11µF	.80mH	33µF
1800	15µF	.27mH	47µF	7.5µF	.50mH	22µF
2000	13µF	.24mH	40µF	6.8µF	.47mH	20mF
3000	8.8µF	.16mH	27µF	4.7µF	.33mH	14µF
4000	6.8µF	.12mH	20µF	3.3µF	.24mH	10µF
6000	4.7µF	82mH	13µF	2.2µF	.27mH	6.8µF
8000	3.3µF	60mH	20µF	1.5µF	.12mH	5.0µF
1000	2.7µF	47mH	8.2µF	1.3µF	.10mH	3.9µF
12000	2.2µF	39mH	6.8µF	1.1µF	82mH	3.3µF

Note: These values are approximate for pure resistive loads only. Speakers vary in impedance with frequency and type, and will not match these values exactly.

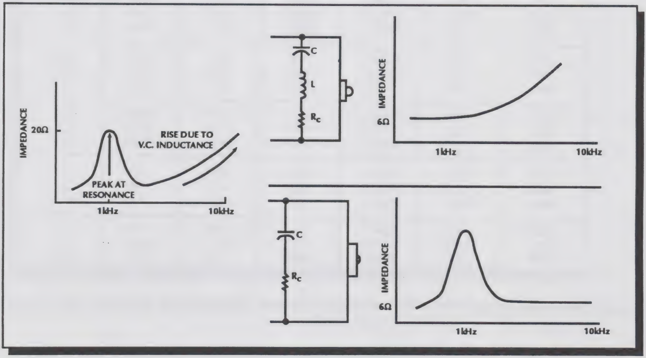
IMPEDANCE COMPENSATION NETWORKS

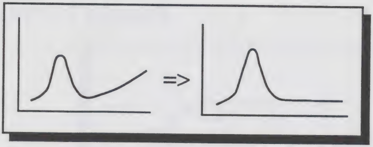
To maximize the operation of passive crossover networks and the performance of a particular driver in a multi-amplifier configuration, impedance compensating networks are very useful.

Passive crossover design is based on a static (non-changing) resistance/impedance. Unfortunately, speakers present a dynamic load (impedance) to a crossover not a static one. Taking measurements with your IM-1 will help uncover possible impedance anomalies. With an accurate impedance plot, you can construct an impedance compensating network that is designed to minimize impedance variations at, or near, the crossover frequency that can ensure that a crossover will behave as it was intended.

In an ***active*** system, the dynamic impedance of a driver is also very important. Many manufacturers compensate for driver anomalies with impedance compensating networks. (Some manufacturers recommend that you only use their passive crossover with their speakers.) While, at Rockford, we do not put such limitations on our speaker systems; many drivers can benefit (sound even better) from impedance networks. Remember, amplifiers produce different amounts of power (watts) depending on the load presented to it. Minimizing impedance fluctuations through the use of impedance networks can bring your sound system to the next level of sonic performance. But, the only way to construct these networks is to measure the impedance response with a device like the IM-1.

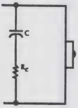
Below are some examples of impedance compensating networks.





Zobel Network

Object: To flatten impedance response rise caused by voice coil inductive interactions (reaction). This improves LP crossover response and reduces harshness in tweeters while providing stable impedance platform for passive attenuation circuits (L-Pad).



$$C = \frac{L_e}{R_C^2}$$

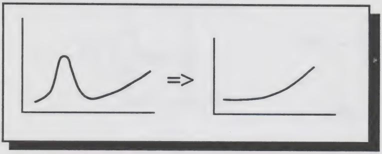
where L_e = Driver's voice coil inductance in Henrys (not millihenrys)

$R_C = 1.25 \times R_E$

R_E = DC resistance of speaker

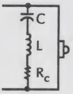
NOTE: This set of values for R & C are approximate. Adjust values to achieve flat impedance response.

NOTE: 1 millihenry or 1mH = 1 x 10^{-3}H



Series Notch Filters

Object: To dampen and eliminate the effect that driver resonances can cause on crossover response.



$$C = \frac{0.1592}{R_e Q_{es} f_s} \; ; \qquad L = \frac{0.1592\,(Q_{es}\,R_e)}{f_s}$$

$$R_C = \left[R_e + \frac{(Q_{es}\,R_e)}{Q_{ms}} \right]$$

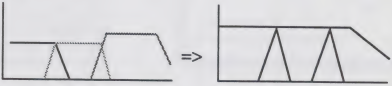
NOTE: Driver parameters available from speaker factbook.

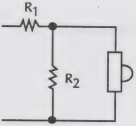
If T/S parameters are unavailable:

$$C = \frac{0.03003}{f} \; ; \quad L = \frac{0.02252}{f^2 c} \; ; \quad R_C = R_e \text{ (DC resistance of speaker)}$$

NOTE: Measure impedance and adjust R_c in 0.5Ω steps to achieve desired impedance response.

L-Pad Network





$$R_2 = \frac{10^{(A/20)} \times Z}{1 - 10^{(A/20)}}$$

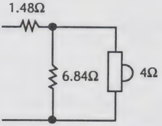
$$R_1 = Z - \left(\frac{1}{R_2} + \frac{1}{Z}\right)^{-1}$$

Where: Z = total driver impedance
A = amount of desired attenuation level in –dB (i.e. –3dB; A = –3)

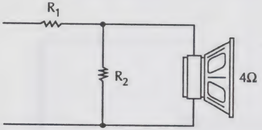
Example:
$Z = 4\Omega$; $A = -4$ (–4dB)

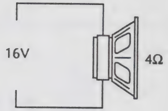
$$R_2 = \frac{10^{(-4/20)} \times 4}{1 - 10^{(-4/20)}} = \frac{10^{(-1/5)} \times 4}{1 - 10^{(-1/5)}} = 6.84\Omega$$

$$R_1 = 4 - \left(\frac{1}{6.84} + \frac{1}{4}\right)^{-1} = 1.48\Omega$$



L-Pad Example





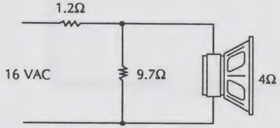
W = 64 Watts

Goal: Reduce acoustical output by 3dB (32 Watts power applied).

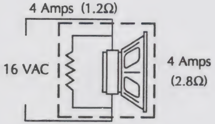
$$R_2 = \frac{10^{(A/20)} \bullet Z}{1 - 10^{(A/20)}} \qquad R_1 = Z - \left(\frac{1}{R2} + \frac{1}{Z}\right)^{-1}$$

$$R_2 = \frac{10^{(-3/20)} \bullet 4}{1 - 10^{(-3/20)}} = 9.696\Omega \approx 9.7\Omega$$

$$R_1 = 4 - \left(\frac{1}{9.696} + \frac{1}{4}\right)^{-1} = 1.168\Omega \approx 1.2\Omega$$

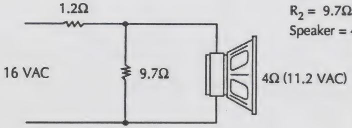


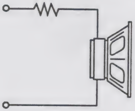
Total Ω = 4 Ohms
Total W = 64 Watts



$I_t = I_1 = I_2 = 4$ Amps

R_1 = 1.2Ω, 4 Amps, 19.2 Watts, 4.8 Volts
R_2 = 9.7Ω, 12.9 Watts
Speaker = 4Ω = 31.4 ≈ 32 Watts





Re = DC resistance of voice coil

$$Rc = \frac{Re\,(1 - 10^{\frac{A}{20}})}{10^{\frac{A}{20}}}$$

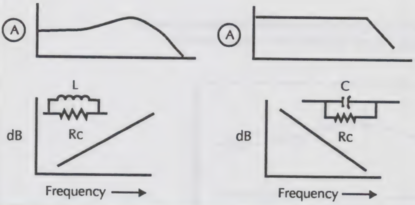
where: R_c = Attenuation resistor
R_e = DC resistance of voice coil
A = dB attenuation i.e., –3dB A = –3

to find dB of attenuation...

$$A\,(dB) = 20 \log \left(\frac{Re}{Re + Rc}\right)$$

Frequency Response Shaping Networks

Contour Networks



Example 1:
Response is rising with increasing frequency

$$L = \frac{0.15916}{f} \qquad R_C = \frac{Re\,(1-10^{\frac{A}{20}})}{10^{\frac{A}{20}}}$$

$$z = \frac{R_C X}{(R^2 + X^2)^{1/2}}$$

L = inductance in Henries
f = starting point of network interactance
z = total circuit impedance
Re = resistance of voice coil
A = amplitiude of signal
–3dB = –3 (for A)

$X_L = 2\pi f L$ formula for reactance of an inductor
f is frequency of maximum desired attenuation
X = component reactance at frequency of maximum attenuation

Example 2:
Response is rising with decreasing frequency

$$c = \frac{0.15916}{f}$$

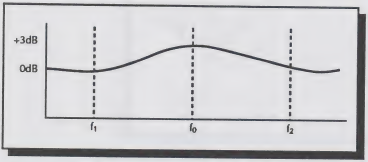
$$z = \frac{RX}{(R^2 + X^2)^{1/2}}$$

c = capacitance in farads
f = starting point of network interactance
z = total circuit impedance

$X_C = \dfrac{1}{2\pi f c}$ formula for reactance of an inductor
f is frequency of maximum desired attenutation
X = component reactance at frequency of maximum attenuation

Parallel Trap Filters

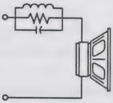
Design Concept: Network designed to compensate for frequency peaks caused by driver response and vehicle acoustics.



Step 1: Find f_0

Step 2: Find f_1 & f_2 (–3dB point referenced to f_0)

Step 3: Design Circuit



$$C = \frac{0.03003}{f_0} \quad \text{in farads}$$

$$L = \frac{0.02252}{f_0^2 C}$$

$$B = \text{–3dB Bandwidth} = (f_2 - f_1)$$

$$R = \frac{1}{6.2832\ CB}$$

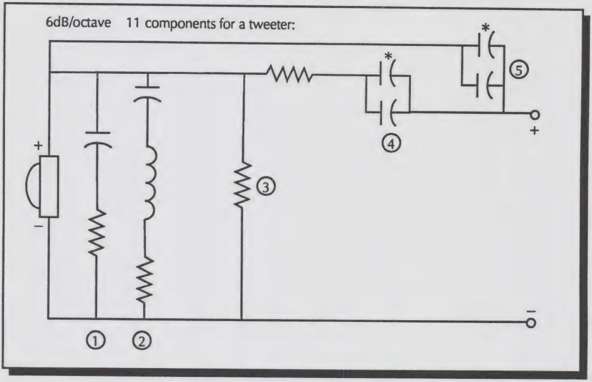
Example: $f_0 = 1000\text{Hz}$; $f_2 = 2000\text{Hz}$; $f_1 = 500\text{Hz}$

$$C = \frac{0.03003}{f_0} = 30.03\mu f$$

$$L = \frac{0.02252}{(1000)^2\,(3.003 \times 10^{-5})} = 0.75\text{mH}$$

$$B = 2000 - 500 = 1500$$

$$R = \frac{1}{6.2832\,(3.003 \times 10^{-5})\,(1500)} = 3.5\Omega$$



Types of passive filters used:

1. **Zobel Network**
   Compensates for impedance rise due to inductance of voice coil.

2. **Series Notch Filter**
   Compensates for impedance rise due to resonance of driver.

3. **L-pad Network**
   Compensates for sensitivity differences between tweeter & midrange.

4. **Crossover**
   High-pass crossover passes high frequencies while blocking low frequency information.

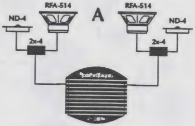
5. **Frequency Shaping Contour Network**
   Network compensates for increasing frequency response roll-off due to off-axis frequency response of the driver.

* Bypass capacitor combines with other capacitor to provide greater performance of a capacitor at a much reduced cost.

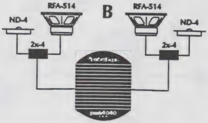
ACTIVE SYSTEMS

- Active System Design
- Building Crossover Cards
- Designing & Using Sub-sonic Filters
- Mapping Sub-sonic Filters

System 1

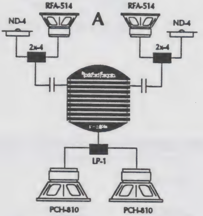


Crossover: 100Hz HP
Products: 60ix
RFA-514

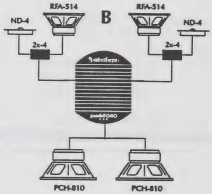


Crossover: 100Hz HP
Products: 4040
RFA-514

System 2

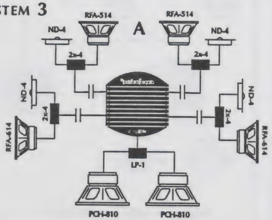


Crossover: 30Hz HP
Products Added: (2) PCH-810
(1) LP-1
(2) 200µf Caps

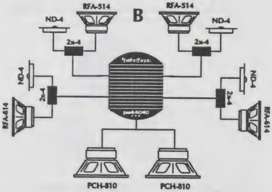


Crossover: CH 1/2 100Hz HP
CH 3/4 100Hz LP
Products Added: (2) PCH-810

System 3



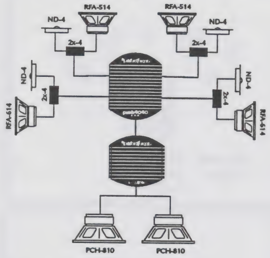
Crossover: 30Hz HP
Products Added: RFA-614
(2) 200µf Caps



Crossover: 100Hz HP; 100Hz LP
Products Added: RFA-614

SYSTEM 4

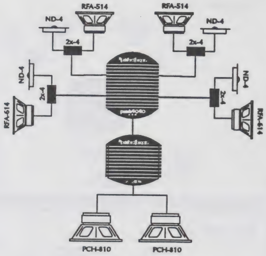
A



Products Added: Punch 4040DSM
Products Not Used: (4) 200µf caps
(1) LP1
Crossover : CH 1/2 100Hz HP
CH 3/4 100Hz HP
*CH S.O. 100Hz LP

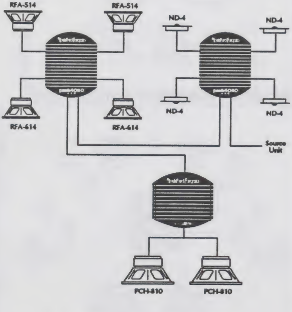
*S.O. is summed output of 4040DSM

B



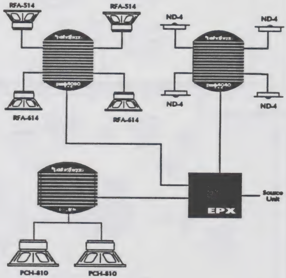
Products Added: Punch 60ix

SYSTEM 5



Products Added: Punch 4080DSM
Products Not Used: (2) 2x-4
Feature Added: Tri-amp Design

SYSTEM 6



Products Added: 3X or EPX
Feature Added: Front Rear Fade

BUILDING CROSSOVER CARDS

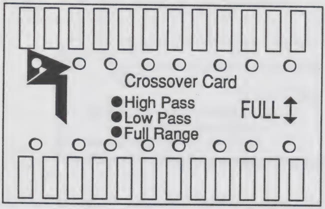


Figure 1

The field programmable module is used in Punch and Series 1 amplifiers as well as Punch electronic crossovers. It consists of a small printed circuit (PC) board that accepts standard 5% and 1% resistors as the method of programming (Figure 1). The value of the resistors on the PC board will determine the crossover point. Connection to the amplifier/crossover motherboard is made via the board edge contacts or "fingers" on one edge of the module. The crossover slope is fixed at 12dB per octave but the alignment can be Butterworth or Bessel or another "Q" value.

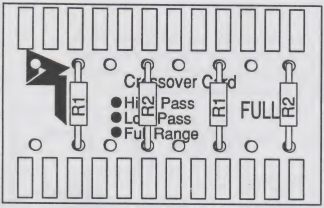


Figure 2

A module that has been programmed with a ***Butterworth*** alignment can be used as a high-pass or low-pass filter ***without*** changing any resistor values. The ***type*** of filter, high-pass or low-pass, is determined by the orientation of the module in the connector on the amplifier/crossover motherboard.

A module that has been programmed with a ***Bessel*** or other Q alignment ***must be programmed specifically as a high-pass or as a low-pass.***

The full range feature is available on every module.

Use the charts on the following pages to select the appropriate resistor values to be placed on modules.

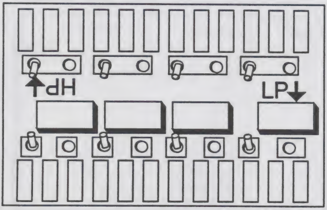


Figure 3

After inserting the resistors (Figure 2), be sure to crimp the leads away from the board edge as shown in Figure 3 after soldering. This will eliminate the possibility of lead ends shorting out to the connector. Also be careful to keep any solder from flowing onto the board edge contacts, otherwise the board will not fit properly into the edge connector. Remove any excess solder from the board edge contacts before attempting to insert the crossover module.

$$\frac{3386}{f_o} = R \text{ (in k}\Omega\text{) for .047}\mu\text{f cap}$$

$$\frac{7234}{f_o} = R \text{ (in k}\Omega\text{) for .022}\mu\text{f cap}$$

The actual formula is: $R = \dfrac{1}{2\pi f_o c}$

Where $R = \Omega$
f_o = desired crossover frequency
c = capacitor in farads
ex: $.047 \times 10^{-6}$ for .047µf cap

Reading Resistors

Resistors impede the flow of current and dissipate electrical energy as heat. Resistors are measured in units called "ohms," represented by the symbol Ω. The fixed resistors look like cylinders with a lead coming out of each end and have several color bands around them. The color bands identify the value of each resistor. For Rockford's crossover cards two types of fixed resistors can be used: metal film resistors and carbon film resistors.

The metal film resistors have five color bands. Figure CI-1 shows the meaning of each band.
The carbon film resistors have four color bands. Figure CI-2 shows the meaning of each band.
Each color band on a resistor represents a number. The color bands represent a code - The Color Code - to identify the value and tolerance of each resistor. Table 1 shows the resistor color code.

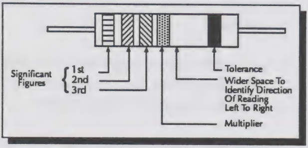


Figure CI-1. Meaning of color bands on metal film resistors.

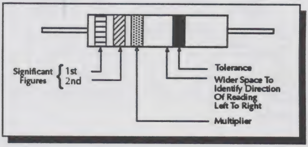


Figure CI-2. Meaning of color bands on carbon film resistors.

Color	lst Digit	2nd Digit	3rd Digit*	Multiplier	Tolerance
Black	0	0	0	1	-
Brown	1	1	1	10	± 1%
Red	2	2	2	100	-
Orange	3	3	3	1000	-
Yellow	4	4	4	10000	-
Green	5	5	5	100000	-
Blue	6	6	6	1000000	-
Violet	7	7	7	10000000	-
Grey	8	8	8	100000000	-
White	9	9	9	1000000000	-
Silver				.01	± 10%
Gold				.1	± 5%

TABLE 1 - Resistor color codes

Electronic Crossover Field Programmable Module Resistor Chart – Butterworth Alignment

Use 5% resistors in conjunction with .022µF standard capacitor.

	Low Pass		High Pass	
Freq.	R1	R2	R1	R2
18.5Hz	390k Ω	390k Ω	390k Ω	390k Ω
26Hz	270k Ω	270k Ω	270k Ω	270k Ω
33Hz	220k Ω	220k Ω	220k Ω	220k Ω
40Hz	180k Ω	180k Ω	180k Ω	180k Ω
48Hz	150k Ω	150k Ω	150k Ω	150k Ω
60Hz	120k Ω	120k Ω	120k Ω	120k Ω
72Hz	100k Ω	100k Ω	100k Ω	100k Ω
88Hz	82k Ω	82k Ω	82k Ω	82k Ω
106Hz	68k Ω	68k Ω	68k Ω	68k Ω
130Hz	56k Ω	56k Ω	56k Ω	56k Ω
154Hz	47k Ω	47k Ω	47k Ω	47k Ω
185Hz	39k Ω	39k Ω	39k Ω	39k Ω
220Hz	33k Ω	33k Ω	33k Ω	33k Ω
270Hz	27k Ω	27k Ω	27k Ω	27k Ω
330Hz	22k Ω	22k Ω	22k Ω	22k Ω
400Hz	18k Ω	18k Ω	18k Ω	18k Ω
480Hz	15k Ω	15k Ω	15k Ω	15k Ω
600Hz	12k Ω	12k Ω	12k Ω	12k Ω
720Hz	10k Ω	10k Ω	10k Ω	10k Ω
880Hz	8.2k Ω	8.2k Ω	8.2k Ω	8.2k Ω
1.06kHz	6.8k Ω	6.8k Ω	6.8k Ω	6.8k Ω
1.3kHz	5.6k Ω	5.6k Ω	5.6k Ω	5.6k Ω
1.54kHz	4.7k Ω	4.7k Ω	4.7k Ω	4.7k Ω
1.85kHz	3.9k Ω	3.9k Ω	3.9k Ω	3.9k Ω
2.2kHz	3.3k Ω	3.3k Ω	3.3k Ω	3.3k Ω
2.7kHz	2.7k Ω	2.7k Ω	2.7k Ω	2.7k Ω
3.3kHz	2.2k Ω	2.2k Ω	2.2k Ω	2.2k Ω
4.0kHz	1.8k Ω	1.8k Ω	1.8k Ω	1.8k Ω
4.8kHz	1.5k Ω	1.5k Ω	1.5k Ω	1.5k Ω
6.0kHz	1.2k Ω	1.2k Ω	1.2k Ω	1.2k Ω
7.2kHz	1k Ω	1k Ω	1k Ω	1k Ω
8.8kHz	820 Ω	820 Ω	820 Ω	820 Ω
10.6kHz	680 Ω	680 Ω	680 Ω	680 Ω
13.0kHz	560 Ω	560 Ω	560 Ω	560 Ω
15.4kHz	470 Ω	470 Ω	470 Ω	470 Ω

Electronic Crossover Field Programmable Module Resistor Chart – Bessel Alignment

Use 5% resistors in conjunction with .022µF standard capacitor.

	Low Pass		High Pass	
Freq.	R1	R2	R1	R2
30Hz	200k Ohm	100k Ohm	470k Ohm	360k Ohm
35Hz	180k Ohm	82k Ohm	390k Ohm	300k Ohm
40Hz	150k Ohm	68k Ohm	330k Ohm	270k Ohm
50Hz	120k Ohm	56k Ohm	270k Ohm	220k Ohm
60Hz	100k Ohm	47k Ohm	240k Ohm	180k Ohm
75Hz	82k Ohm	39k Ohm	200k Ohm	160k Ohm
90Hz	68k Ohm	33k Ohm	160k Ohm	130k Ohm
106Hz	56k Ohm	27k Ohm	130k Ohm	100k Ohm
130Hz	47k Ohm	22k Ohm	110k Ohm	82k Ohm
160Hz	39k Ohm	18k Ohm	91k Ohm	75k Ohm
190Hz	30k Ohm	15k Ohm	75k Ohm	62k Ohm
250Hz	24k Ohm	12k Ohm	56k Ohm	43k Ohm
300Hz	20k Ohm	10k Ohm	47k Ohm	36k Ohm
350Hz	18k Ohm	8.2k Ohm	39k Ohm	30k Ohm
425Hz	15k Ohm	6.8k Ohm	33k Ohm	27k Ohm
520Hz	12k Ohm	5.6k Ohm	27k Ohm	22k Ohm
610Hz	10k Ohm	4.7k Ohm	24k Ohm	18k Ohm
740Hz	8.2k Ohm	3.9k Ohm	20k Ohm	16k Ohm
870Hz	6.8k Ohm	3.3k Ohm	16k Ohm	13k Ohm
1.1kHz	5.6k Ohm	2.7k Ohm	13k Ohm	10k Ohm
1.3kHz	4.7k Ohm	2.2k Ohm	11k Ohm	8.2k Ohm
1.6kHz	3.9k Ohm	1.8k Ohm	9.1k Ohm	7.5k Ohm
1.9kHz	3.0k Ohm	1.5k Ohm	7.5k Ohm	6.2k Ohm
2.5kHz	2.4k Ohm	1.2k Ohm	5.6k Ohm	4.3k Ohm
3.0kHz	2.0k Ohm	1.0k Ohm	4.7k Ohm	3.6k Ohm
3.5kHz	1.8k Ohm	820 Ohm	3.9k Ohm	3.0k Ohm
4.25kHz	1.5k Ohm	680 Ohm	3.3k Ohm	2.7k Ohm
5.2kHz	1.2k Ohm	560 Ohm	2.7k Ohm	2.2k Ohm
6.1kHz	1.0k Ohm	470 Ohm	2.4k Ohm	1.8k Ohm
7.4kHz	820 Ohm	390 Ohm	2.0k Ohm	1.6k Ohm
8.7kHz	680 Ohm	330 Ohm	1.6k Ohm	1.3k Ohm
11kHz	560 Ohm	270 Ohm	1.3k Ohm	1.0k Ohm

SUB–SONIC FILTERS & VARIABLE Q CROSSOVERS

Formula For Variable Q High Pass Crossover

$$R_1 = \left(\frac{1.81 \times 10^6}{f_o}\right)\left(\frac{1}{Q} + \sqrt{\left(\frac{1}{Q}\right)^2 + 4.72}\right)$$

$$R_2 = \frac{52.3 \times 10^{12}}{(R_1)(f_o^2)}$$

NOTE: R_1 and R_2 values for use with .022µf capacitors.

R_1 and R_2 = resistors on crossover module.
f_o = desired frequency point
Q = Q response at roll-off

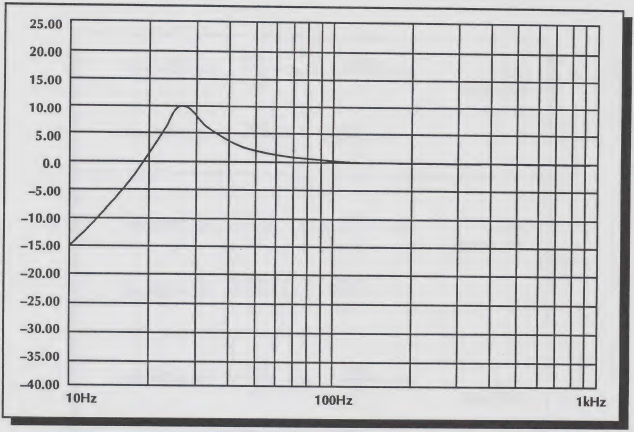
For crossover cards with .047µf capacitors:

$$R_1 = \left(\frac{8.51 \times 10^5}{f_o}\right)\left(\frac{1}{Q} + \sqrt{\left(\frac{1}{Q}\right)^2 + 4.72}\right)$$

$$R_2 = \frac{1.16 \times 10^{13}}{(R_1)(f_o^2)}$$

or take results from formula for .022µf and take results R_1 and R_2 and multiply results by 0.47 to obtain value for use with crossover modules using .047µf capacitors.

CROSSOVER AMPLITUDE (dB) VS. FREQUENCY (Hz)



Example:

Design = Q of 3.5 approximate boost at crossover frequency = 10dB
Capacitor Value = .022µf
Desired f_o = 25Hz

$$R_1 = \left(\frac{1.81 \times 10^6}{f_o}\right)\left(\frac{1}{Q} + \sqrt{\left(\frac{1}{Q}\right)^2 + 4.72}\right)$$

$$= \left(\frac{1.81 \times 10^6}{25}\right)\left(\frac{1}{3.5} + \sqrt{\left(\frac{1}{3.5}\right)^2 + 4.72}\right) = 179{,}333\Omega \text{ or } 179.3\text{k}\Omega$$

$$R_2 = \frac{52.3 \times 10^{12}}{(R_1)(f_o)^2} \Rightarrow \frac{52.3 \times 10^{12}}{(179{,}333)(25)^2} = 466{,}617\Omega \text{ or } 466.6\text{k}\Omega$$

NOTE: At lower frequencies, crossover response is improved with the use of .047µf capacitors.

POCKET CALCULATOR SOLUTIONS

(All Resistor Values in kΩ)

Q3.5 (10dB boost)	$R1 = \frac{4480}{f_o}$; $R2 = \frac{11674}{f_o}$	(for .022µf cap.)
	$= \frac{2107}{f_o}$; $= \frac{5487}{f_o}$	(for .047µf cap.)
Q2.0 (6dB boost)	$R1 = \frac{4940}{f_o}$; $R2 = \frac{10587}{f_o}$	(for .022µf cap.)
	$= \frac{2322}{f_o}$; $= \frac{4976}{f_o}$	(for .047µf cap.)
Q1.4 (3dB boost)	$R1 = \frac{5432}{f_o}$; $R2 = \frac{9628}{f_o}$	(for .022µf cap.)
	$= \frac{2553}{f_o}$; $= \frac{4525}{f_o}$	(for .047µf cap.)

Example:

Q = 3.5 F = 25Hz

$$R1 = \frac{4483}{25} = 179.3k\Omega;\ R2 = \frac{11665}{25} = 466.6k\Omega$$

DRIVER DATA

Name: ROCKFORD FOSGATE RFA-124

Fs	=	21.0	Pe	=	31.0
Qts	=	0.310	Xmax	=	0.360
Vas	=	9.888	Ref	=	89.00
Z	=	4	Diam	=	12.00
Nd	=	1	Lt	=	——

Enclosure Preference

S P

PORTED

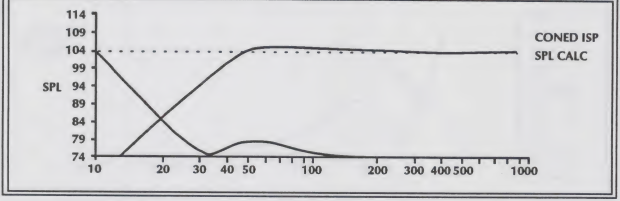
<PREV>
SPL CALC
CONED ISP
D SEARCH
NEW DRVR
GRID TOG
CROSHAIR
AXIS RES
OVLY TOG
HARDCOPY

PORTED HIGHPASS [4+0]

Type: FORCED – V b SPECIFIED

Volume	[Vb]	=	2.000	ft^3
Tuning	[Fb]	=	34.47	Hz
Cutoff	[F3]	=	42.42	Hz
Ripple	[R]	=	2.089	dB

[ ] Electrical Overlay



DRIVER DATA

Name: ROCKFORD FOSGATE RFA-124

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Qts	=	0.310	Xmax	=	0.360
Vas	=	9.888	Ref	=	89.00
Z	=	4	Diam	=	12.00
Nd	=	1	Lt	=	——

Enclosure Preference

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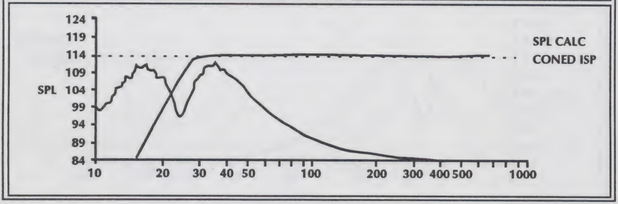
D SEARCH
NEW DRVR
GRID TOG
CROSHAIR
AXIS RES
OVLY TOG
HARDCOPY

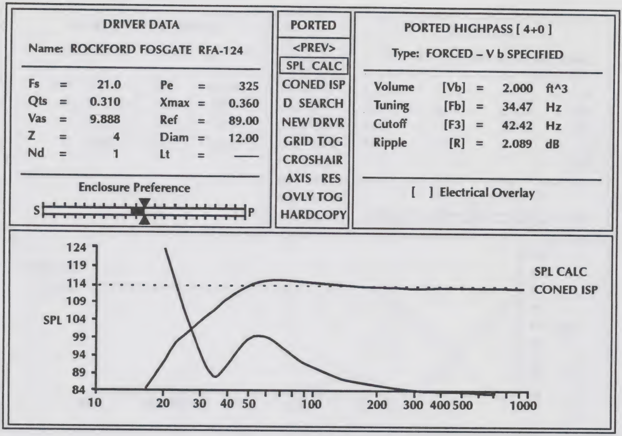
PORTED HIGHPASS [4+0]

Type: USER SPECIFIED Vb & Fb

Volume	[Vb]	=	2.000	ft^3
Tuning	[Fb]	=	25.50	Hz
Cutoff	[F3]	=	42.42	Hz
Ripple	[R]	=	–.528	dB

[■] Electrical Overlay

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Qts = 0.310
Vas = 9.888
Z = 4
Nd = 1
Pe = 325
Xmax = 0.360
Ref = 89.00
Diam = 12.00
Lt = ——
Enclosure Preference
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SPL CALC
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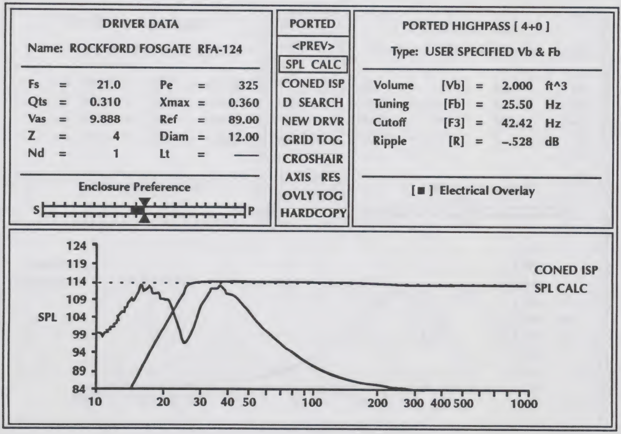
D SEARCH
NEW DRVR
GRID TOG
CROSHAIR
AXIS RES
OVLY TOG
HARDCOPY
PORTED HIGHPASS [4+0]
Type: FORCED – V b SPECIFIED
Volume [Vb] = 2.000 ft^3
Tuning [Fb] = 34.47 Hz
Cutoff [F3] = 42.42 Hz
Ripple [R] = 2.089 dB
[ ] Electrical Overlay
SPL
124
119
114
109
104
99
94
89
84
10
20
30
40
50
100
200
300
400
500
1000
SPL CALC
CONED ISP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Fs = 21.0
Qts = 0.310
Vas = 9.888
Z = 4
Nd = 1
Pe = 325
Xmax = 0.360
Ref = 89.00
Diam = 12.00
Lt = ——
Enclosure Preference
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PORTED
<PREV>
SPL CALC
CONED ISP
D SEARCH
NEW DRVR
GRID TOG
CROSHAIR
AXIS RES
OVLY TOG
HARDCOPY
PORTED HIGHPASS [4+0]
Type: USER SPECIFIED Vb & Fb
Volume [Vb] = 2.000 ft^3
Tuning [Fb] = 25.50 Hz
Cutoff [F3] = 42.42 Hz
Ripple [R] = –.528 dB
[■] Electrical Overlay
SPL
124
119
114
109
104
99
94
89
84
10
20
30
40
50
100
200
300
400
500
1000
CONED ISP
SPL CALC

DRIVER DATA

Name: ROCKFORD FOSGATE RFA-124

Fs	=	21.0	Pe	=	350
Qts	=	0.310	Xmax	=	0.360
Vas	=	9.888	Ref	=	89.00
Z	=	4	Diam	=	12.00
Nd	=	1	Lt	=	——

Enclosure Preference

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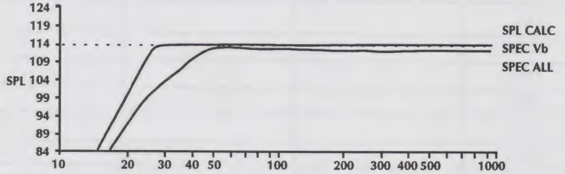
SPL CALC
CONED ISP
D SEARCH
NEW DRVR
GRID TOG
CROSHAIR
AXIS RES
OVLY TOG
HARDCOPY

PORTED HIGHPASS [4+0]

Type: FORCED – V b & Fb

Volume	[Vb]	=	2.000	ft^3
Tuning	[Fb]	=	26.00	Hz
Cutoff	[F3]	=	42.42	Hz
Ripple	[R]	=	–.359	dB

[■] Electrical Overlay



DRIVER DATA

Name: ROCKFORD FOSGATE RFA-124

Fs	=	21.0	Pe	=	200
Qts	=	0.310	Xmax	=	0.360
Vas	=	9.888	Ref	=	89.00
Z	=	4	Diam	=	12.00
Nd	=	1	Lt	=	——

Enclosure Preference

S P

PORTED

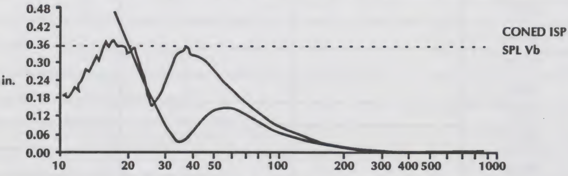
<PREV>
SPL CALC
CONED ISP
D SEARCH
NEW DRVR
GRID TOG
CROSHAIR
AXIS RES
OVLY TOG
HARDCOPY

PORTED HIGHPASS [4+0]

Type: FORCED - VB SPECIFIED

Volume	[Vb]	=	2.000	ft^3
Tuning	[Fb]	=	34.47	Hz
Cutoff	[F3]	=	42.42	Hz
Ripple	[R]	=	2.089	dB

[ ] Electrical Overlay



High-Pass Variable Q Crossover Worksheet
For Data Input into Term-Pro

Capacitor 1 = Resistor 1 =
Capacitor 2 = Resistor 2 =

Notes: ____

____

Frequency in Hertz	Voltage In (Measured AC voltage input to crossover)	Voltage Out (Measured AC voltage output from crossover)	Δ dB (Calculated difference in dB between input and output voltage)
11Hz			
12Hz			
13Hz			
14Hz			
16Hz			
18Hz			
19Hz			
22Hz			
24Hz			
26Hz			
29Hz			
32Hz			
36Hz			
40Hz			
44Hz			
49Hz			
54Hz			
60Hz			
66Hz			
74Hz			
81Hz			
90Hz			
98Hz			

Variable Q Crossover Worksheet Guideline

Step 1

Frequency (Measured in Hertz)

This is the test frequency from your source output. Frequencies from 10Hz to 98Hz are available from your Autosound 2000 Test CD #101. For frequencies 20Hz and above, use your OSC2. The frequencies listed on the worksheet correspond to the input frequencies on your Term-Pro Computer aided design program.

Step 2

Voltage In (Measured AC voltage input to Crossover)

This is the AC voltage from your source unit to the crossover. Select a frequency corresponding to a frequency on the worksheet and measure the AC voltage with an AC voltmeter before the crossover.

Step 3

Voltage Out (Measured AC voltage output from the Crossover)

This is the AC voltage output from the crossover. After measuring the AC voltage input to the crossover, measure the AC voltage output from the crossover that would normally go to the amplifiers.

Step 4

Δ dB (Calculated difference in dB between the input and output voltage)

This is the difference in voltage in dB between the input and output voltage. This is the transfer function of the high Q, high-pass filter. To calculate the difference in dB, use the following formula.

$$\Delta \text{ dB} = 20 \log \left[\frac{\text{Output Voltage from Crossover}}{\text{Input Voltage to Crossover}} \right]$$

TEST EQUIPMENT

• PD1/PD2
PHASE DETECTOR AND
POSITIVE PULSE GENERATOR

• IM-1
IMPEDANCE METER

• OSC2
PINK NOISE / SINE WAVE GENERATOR

• CD #101
LOW FREQUENCY TEST CD

PD1 / PD2

Feature	Procedure	Benefit
Determine speaker/wire orientation	Use PD2 to pulse speaker wires and verify speaker location	Safe method of testing location of speakers in vehicle
Determine OEM wiring	Using PD1/PD2 to find the location and polarity of OEM speakers and their wiring	Simple, safe method of determining proper wiring configuration and speaker placement in an OEM system
Determine polarity of speakers in a system	Using PD1/PD2 to determine polarity of speakers in system through passive crossovers, etc.	Safe, easy method of proper system wiring and best sonic performance
Determine absolute polarity of a system	Using CD101 (track #3), measure polarity of system from source unit to speaker	Only method to determine absolute polarity of a system for best sonic performance

IM 1

Feature	Procedure	Benefit
Troubleshoot system	Measure impedance of the speaker system	Ensure that system has a well behaved impedance response
Measure impedance of low frequency system	Plot impedance to determine the best wiring configuration for maximizing amplifier output (series or parallel connections or a combination of both)	Optimizes use of power in a system
Measure impedance of a speaker	Plot impedance of a speaker and build compensating networks to smooth impedance response	Provides stable impedance for well behaved crossover response

OSC2		
Feature	**Procedure**	**Benefit**
Find tuning frequency of a vented enclosure	Measure actual tuning frequency of enclosure and adjust to desired tuning frequency	Ensures maximum performance from a vented system
Troubleshoot system	Check and verify operation of equipment by substituting known working isolated source	Easy method of checking operation of all components, cable and connections
Troubleshoot system noise	Substitute component in question with known working, noise free, isolated source	Easy method for verifying noise source in system
Find resonances of vehicle	Sweep the vehicle with sine wave output and find the frequencies that cause rattles and buzzes and damp these unwanted noises	Quick, easy method of locating and eliminating unwanted noises in vehicle and system
Find peak resonance of system	Sweep vehicle with sine wave output and find frequency that system will have highest (most efficient) output	Help select software that will produce the highest sound pressure level for the vehicle/ system
Adjust RTA (Real Time Adjustment)	Using pink noise output of OSC2, measure frequency response of system, make adjustment to the system and note the changes in response	Easy method of measuring and adjusting the system for smooth octave to octave spectral balance
Measure transfer function of vehicle	Plot frequency response of vehicle with test enclosure and subtract near-field response of test enclosure from measurement	Inputting transfer function of the vehicle into Term-Pro will provide user with more accurate box response predictions

CD101		
Feature	**Procedure**	**Benefit**
Set gain controls of system	Use track 99 (all bits high) on CD to adjust system so that maximum input signal will not clip electronic components of system	This process will provide maximum signal to noise ratio, minimum reliability of all components in system
Determine tuning frequency of vented system	Using tracks 10-98, measure actual tuning frequency of enclosure and adjust to desired tuning frequency	Ensures maximum performance from a vented system
Find resonance of vehicle	Using tracks 4, 5, 10-98 to sweep the vehicle, find what frequencies cause rattles and buzzes then damp the unwanted noises	Quick, easy method of locating and eliminating unwanted noises in vehicle and system
Find peak resonance of system	Using tracks 10-98, sweep vehicle with sine wave output and find frequency that system will have highest (most efficient) output	Help select software that will produce the highest sound pressure level for the vehicle/ system
Determine absolute polarity of a system	Using track 3 and PD1, measure polarity of system from source unit to speaker	Only method to determine absolute polarity for best sonic performance
Determine F3 of a low frequency system	Use tracks 10-98 to set system level to medium output and locate the frequency where the system's output reduces drastically	Use F3 to design and install a sub-sonic filter that will limit output and excursion below speakers operational bandwidth reducing distortion and improving output capability
Determine sonic characteristics of low frequency system	Using track 9, listen to the attack of the low frequency system (a dull sounding attack indicates poor transient response) small improvements mounting tuning and adjustment are very noticeable	Easy way to determine and maximize the transient performance of a low frequency system

PD1 / PD2

Phase Detector and Positive Pulse Generator

- Speaker / Wire Configuration
- OEM Wiring Configuration
- Speaker Polarity
- Absolute Polarity (in a System)

DETERMINING DRIVER LOCATION AND POLARITY

When installing an audio system, two very important factors determine its sonic potential: ***Location*** and ***Polarity***.

Location is obvious. Selecting the proper location and wiring the driver in that location is a major factor in the system's ability to recreate a sound stage. **Polarity** is less understood. Most people can understand that if two speakers (a left channel and right channel midrange) are out of polarity, constructive/destructive interference will degrade the performance of the system.

What is equally important is the absolute polarity of the system. Software (the music we listen to) was recorded with the understanding that the end user's system would reproduce the information as it was recorded. Some may argue that as long as the relative polarity between drivers is correct, the result will be the same. A simple experiment with the PD2 will demonstrate this is not the case.

To compensate for phase shift caused by passive crossovers, manufacturers invert the polarity of one of the drivers (usually the high frequency driver). The driver's terminals will be incorrectly marked to simplify the installation for the installer. If you decide to upgrade to a fully active system, the terminal markings would be incorrect. Relying on a manufacturer to properly mark drivers could result in a compromise that would adversely affect the performance of a system. Determining polarity of a driver can only be done by testing. Listed below are some examples in determining driver location and polarity.

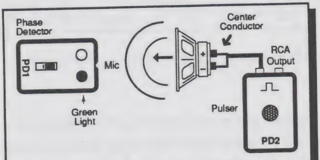
Test Battery and Visual Inspection

Advantage:	Easy to use (only if driver is visible)
Disadvantage:	Harmful to drivers. (The smaller the driver, the greater the possibility of damage)
	Only can determine driver polarity if visible
	No way to determine absolute polarity of system

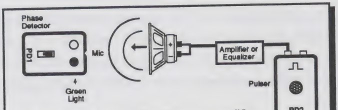
PD1 / PD2 (with the use of Test CD #101)

Advantage:	Easy to use
	Safe for all drivers
	Senses driver pressure to determine polarity
	Determine absolute polarity of system when used with CD #101
Disadvantage:	Only available from ROCKFORD (Disadvantage to other dealers)

PD2 output will drive either an amplifier/preamp signal or a speaker directly.

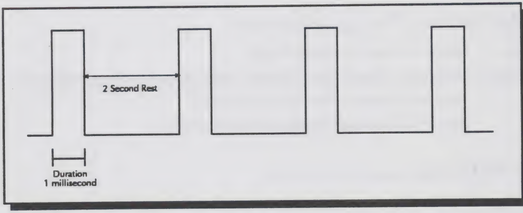


Using the Pulser to Directly Drive a Speaker with Positive Pulses



Using the Pulser to Supply Positive Pulses to an Amplifier or Equalizer for Polarity Checking.

Output from PD2 (from .1V to 4V @ 4Ω) is a **Square Wave Output.**

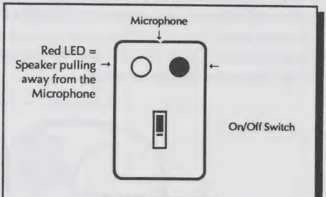


PD1 Phase Detector
PD2 Positive Pulse Generator

The PD1 Phase Detector is a small, handheld, battery operated diagnostic tool designed to quickly and accurately determine the relative phase of a speaker or a speaker system. There is an internal microphone in the PD1's front panel and two top-mounted LEDs (light emitting diodes) which indicate the following:

GREEN = Speaker cone is **pushing towards** the PD1's microphone or

RED = Speaker cone is **pulling away** from the PD1's microphone.

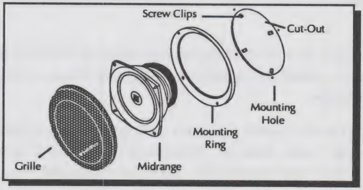


PHASE DETECTOR

The only control on the device is the on/off rocker switch which should be kept in the off position when not in use.

Background

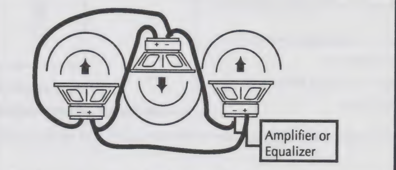
It was long ago determined that sound waves travel through the air at around 1130 feet per second. In the last 100 years we have been creating sound by forcing speakers to move back and forth in the air. (Speakers cannot work in the vacuum of space). The actual motion of a speaker's cone is the result of varying electrical currents flowing in the speaker's voice coil. This voice coil is typically constructed of a double winding of a small gauge solid enameled copper wire. A magnetic field is placed in and around the voice coil by way of a permanent magnet.



The direction in which the electrical current is flowing through the voice coil determines if the speaker's cone will move forward or backward. Conventionally, if a small direct current from a flashlight battery is made to flow through the voice coil and the speaker's cone moves forward, then the positive (+) terminal of the voice coil is the one connected to the positive (+) electrode of the battery.

If the direction of the current is changed by simply reversing the connections to the voice coil, then the **POLARITY** of the speaker is reversed. Also, the **PHASE** of the audio information produced by the speaker is changed by 180°.

If one or more speakers in a speaker system is "out-of-phase" with the rest of the system, then the entire system will suffer the consequences of **CANCELLATION**. Cancellation disrupts the air by smoothing out the sound waves. One out-of-phase speaker can effectively cancel the sound from several in-phase speakers. It is extremely important for the polarity of each voice coil in the sound system to be wired "in-phase" with all the other voice coils in the system.



Cancellation due to Inverse Speaker Polarity has the same Effect as Physically Reversing a Speaker. The Middle Speaker above is Out-Of-Phase with the Others.

COMPLICATIONS

Most speaker manufacturers place a red mark or a (+) on the positive speaker terminal. Some speaker manufacturers place the mark on the negative terminal! Most audio components do not invert the signal as it passes through its circuits, but some components do invert the audio!

Additionally, crossovers have the ability to produce phase differences between their outputs. These differences can have serious consequences. Let's examine some typical crossovers and their input and output phase relationships.

The **ORDER** of a filter is a measure of its rate of attenuation. An **OCTAVE** is a doubling or halving in frequency. For instance, a 400Hz tone is one octave above a 200Hz tone, but a 100Hz tone is one octave below that same 200Hz tone.

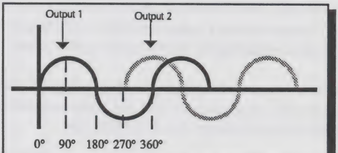
Passive filters are series and/or parallel configured capacitors, inductors, and/or resistors constructed to function as a crossover. Active filters are electronic circuits utilizing op-amps, power supplies and sophisticated circuitry for crossover networks. Sharp cutoff rates and low-losses are features of active crossover networks.

A **CROSSOVER** is a filtering device that either passes or rejects certain designated frequencies. There are two main qualities of a crossover: its order and its "Q". The "Q" of a filter is the sharpness of the filter's cutoff near the resonant frequency where the filtering action just begins to take effect.

A first order network, called a 6dB/octave crossover, will produce a 90° **PHASE SHIFT** between its high and low outputs. This results in the delivery of both constant voltage and constant power through the crossover region. An advantage of the first order network is **ALL PASS**, which means that the sum of the output is identical to that of the input. Since first order networks have a very low rate of attenuation they can harm tweeters and mid-range speakers.

Second order crossovers, also called 12dB/octave networks, produce a 180° phase shift between their outputs. A typical application of a second order network would call for reversing the polarity of the higher frequency driver so that its ouput will be inverted. Now the tweeter will be in phase with the mid-range or woofer. Did the manufacturer of the 12dB/octave crossover already account for this polarity reversal when marking his device?

Third order crossovers, also called 18dB/octave networks, are seldom found in home speakers, but are commonly used in autosound applications. The outputs of the 18dB/octave crossover will be 270° out of phase. Typically the higher frequency speaker's polarity is reversed, yielding a 90° phase difference.



270° Phase Shift Between Outputs of an 18dB/octave Crossover

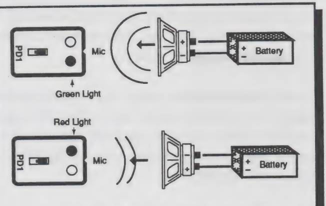
Fourth order crossovers, also called 24dB/octave networks, will keep both outputs in-phase, however, the roll-off associated with these crossovers is very sharp.

Higher order crossover networks follow the same pattern (i.e. 5th=450° difference, 6th=540°, 7th=630°s, etc.). Note: Multiples of 360° can be subtracted from actual phase differences of higher order crossovers.

In general, it is very important to make sure that the phase of each and every speaker in a sound system is correct. Consideration for the phase shifts produced by the crossover network, as well as the speaker enclosure, must be taken into account when designing sound systems.

Operation of PD1

To use the PD1 to determine the polarity of a speaker, the speaker will first have to be driven, or pulsed, in a definite direction. For example, let's take a small flashlight battery and quickly connect the positive electrode to the + or red terminal of a test speaker. Connect the negative electrode to the - terminal of the same test speaker. The speaker cone should move out and forward when it is pulsed in this manner.



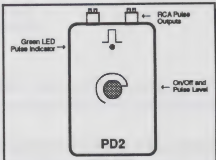
Verifying Polarity with the PD1 Phase Detector

When using the PD1 Phase Detector, try to keep all background noise to a minimum level. The microphone is very sensitive and will respond to outside audio interference.

Position the PD1 about one foot in front of the test speaker and make sure that the microphone is aimed directly at the test speaker. Using the battery as described above, quickly pulse the test speaker. The GREEN LED should momentarily light indicating the speaker was positively pulsed.

Now reverse the battery's leads and negatively pulse the speaker. The RED LED should momentarily light indicating the negative pulse was detected by the PD1. For demonstration purposes, a grille cloth can be draped over the test speaker illustrating the fact that the PD1 can "see through grilles."

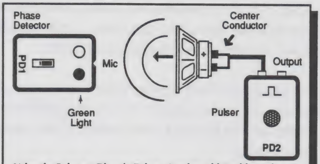
Using a battery to drive a speaker can harm the speaker by causing excessive direct current to flow through the voice coil. Once the voice coil is deformed, the speaker is essentially ruined. A better method of pulsing a speaker would be to use the PD2 Pulser. This battery operated and portable device will provide perfect positive pulses to drive a speaker or an amplifier/equalizer.



PULSER

Using the PD2 Pulser

The PD2 Pulser provides a dual mono audio, positive output pulse approximately .001 seconds in duration. The pulses are spaced about 2 seconds apart to allow the speaker to come to rest. The female RCA plugs on the front of the PD2 are output ports and can pulse a speaker when the sensitivity adjustment is at its maximum level (4V peak) into a 4 Ohm speaker. When the sensitivity knob is set someplace below its maximum level, then the PD2 can provide a pre-amp level positive pulse to drive an amplifier, electronic crossover, equalizer, etc.



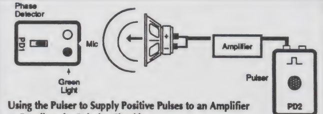
Using the Pulser to Directly Drive a Speaker with Positive Pulses

To demonstrate the function of the PD2 Pulser when used as a positive pulse source for the PD1 Phase Detector, position the phase detector about one foot in front of a test speaker. Connect the output of the PD2 Pulser to the test speaker's terminals and turn the sensitivity control clockwise to the maximum level. The center conductor of the RCA plug is the positive output and directly corresponds to the positive electrode of the battery. When wired so that the inner conductor of the RCA is connected to the + or red terminal of the speaker, then the speaker should pulse TOWARDS the phase detector. Sharp clicks should now be heard in the test speaker. Turn on the PD1 Phase Detector and the GREEN light should be pulsing with the speaker clicks.

Now reverse the leads on the terminals. The RED light should illuminate, indicating that the speaker is pulling away from the microphone of the PD1. The PD2 Pulser can be used in this manner to safely provide positive pulses to sub-woofers, woofers, mid-range speakers and tweeters.

Pre-amp Level Pulsing

To use the PD2 Pulser to drive through an amplifier, equalizer, or electronic crossover, simply connect the PD2 to the input of the component to be pulsed with male-to-male RCA cables. Although the PD2 has two bridged female RCA plug outputs, it may be desirable to use a single RCA cable and move it between the various amplifier channels in the sound system.



Using the Pulser to Supply Positive Pulses to an Amplifier or Equalizer for Polarity Checking.

After the RCA cable(s) are connected between the pulser and the component, activate the amp/eq/xover and then slowly turn the knob clockwise until the pulses can just be detected in the speakers. Next, turn on the PD1 Phase Detector and, with its microphone aimed directly at each speaker to be tested, check the polarity of each speaker.

If many speakers are installed in a vehicle, make sure to check only one speaker at a time by placing the PD1 directly in front of that speaker. To ensure accurate measurements, disconnect all speakers except the one speaker to be tested.

Take care when pulsing tweeters with the PD2. There is no need to overdrive any speaker. The phase detector is very sensitive and if the pulse is audible, the PD1 will determine its polarity.

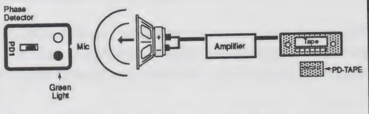
Other Pulsing Methods

THE PD-TAPE is a stereo (two channel) audio cassette tape that was individually mastered to provide positive pulses at the output of a tape player -- even through the tape equalization. The PD-TAPE was recorded for approximately three minutes on each side.

To use the PD-TAPE Positive Pulse Cassette Tape with the PD1 Phase Detector, simply turn on the sound system, insert the tape into the cassette mechanism, and adjust the volume of the system until the positive clicks can just be heard in the speakers. Now, position the PD1 directly in front of each speaker and check the polarity.

NOTE: WHEN USING THE PD-TAPE IN AUTO-REVERSE TAPE DECKS, THE PULSES WILL ONLY BE POSITIVE IN ONE DIRECTION OF TAPE TRAVEL. WHEN THE TAPE REVERSES, THE POLARITY OF THE PULSES WILL ALSO REVERSE.

This means that auto-reverse decks invert the audio in one direction of travel. The test tape will produce all GREEN lights on one direction of travel, but as soon as the tape reverses, only RED lights should blink on the PD1.



Using the PD-TAPE Positive Pulse Cassette Tape

The PD-TAPE can be used as a portable pulse source when used in a portable cassette player such as a Walkman-type device fitted with RCA connectors. The battery-operated portable cassette player can then be moved from car to car and connected in a manner similar to the PD2 Pulser.

At the time of this writing, positive pulses are being mastered onto a demonstration CD (compact disc). The use of the CD demo disc is similar to that of the test tape, except that the CD will only produce positive pulses.

IM-1

Impedance Meter

- System Troubleshooting
- Amplifier Power Optimization
- Impedance Optimization

MEASURING IMPEDANCE

Digital Volt Meter (DVM)

Advantage: Easy to use

Disadvantage: Only measures DCR (Direct Current Resistance)

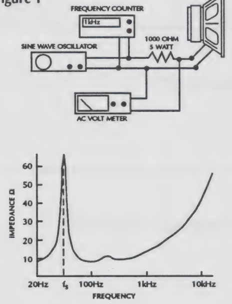
Voltage Divider Method (*See Figure 1*)

Advantage: Accurate measurement of impedance versus frequency

Disadvantage: Expensive, user has to purchase multiple components

Difficult to accomplish testing in a vehicle

Figure 1



Note: Rise due to inductance of voice coil.

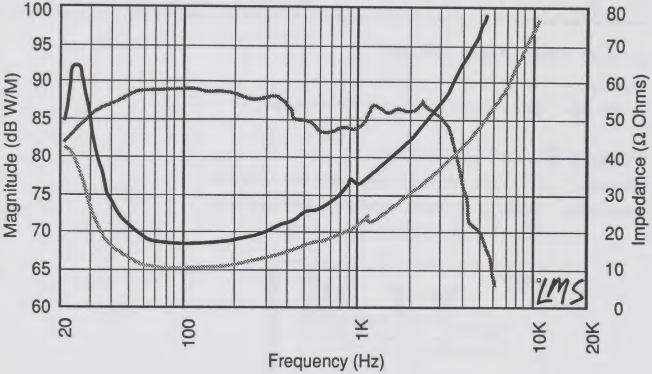
IM-1

Advantage: Accurate measurement of impedance versus frequency

Portable, unit is self-powered

Easy to operate

Disadvantage: Only available from ROCKFORD (Disadvantage to other dealers)



RFA-128 Impedance Plot

Frequency Response Plot for both RFA-124 and RFA-128

RFA-124 Impedance Plot

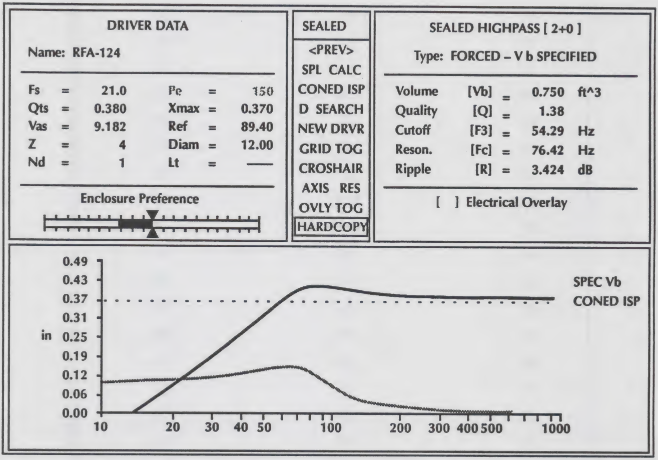
LEGEND

Impedance (Ω), Magnitude (dB W/m), versus Frequency Plot. The graph above depicts the response characteristics of the Audiophile RFA-124 and RFA-128. The impedance plots for both the 4Ω and 8Ω woofers are obtained by measurement while the driver is hanging from a string. The frequency response is the infinite baffle response of the speaker system.

Question:

How can I get a Punch 40i to play as loud as a Punch 200ix?

Using 4 RFA-124s each having a 3/4 cu. ft. (12.24 l) sealed enclosure.
Notice the excursion at 150 Watts.



IMPEDANCE VERSUS FREQUENCY

SPEAKER: **Rockford Fosgate RFA-124**

ENCLOSURE: **3/4 cubic foot sealed enclosure**

FREQUENCY (In Hertz)	IMPEDANCE (In Ohms)	OUTPUT WATTAGE (From one channel of a Punch 200ix)
15	4.2	148
20	4.7	132
25	5.1	122
30	5.6	111
35	6.7	93
40	9.1	74
45	12.0	56
50	14.0	48
55	12.5	54
60	9.2	73
65	7.7	84
70	6.3	99
75	5.9	105
80	5.5	113
85	5.3	117
90	5.2	120
95	5.1	122
100	5.1	122

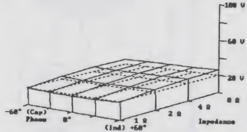
$$\frac{(\text{VOLTAGE})^2}{\text{IMPEDANCE}} = \text{OUTPUT WATTAGE}$$

POWER COMPARISON

AudioGraph – The PowerCube

Amplifier : PUNCH 200ix 14.4V x 2 — Rated Power: 100 W at 4 Ohms
Serial No :
Owner : ROCKFORD CORPORATION

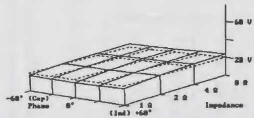
8Ω	–60°	84 W
	–30°	84 W
	0°	83 W
	30°	84 W
	60°	85 W
4Ω	–60°	159 W
	–30°	154 W
	0°	153 W
	30°	154 W
	60°	158 W
2Ω	–60°	266 W
	–30°	251 W
	0°	245 W
	30°	248 W
	60°	261 W
1Ω	–60°	378 W
	–30°	343 W
	0°	333 W
	30°	339 W
	60°	373 W



AudioGraph – The PowerCube

Amplifier : PUNCH 200ix 14.4V x 2 — Rated Power: 100 W at 4 Ohms
Serial No :
Owner : ROCKFORD CORPORATION

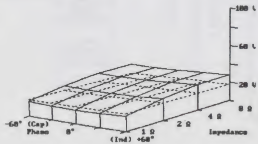
8Ω	–60°	70 W
	–30°	70 W
	0°	69 W
	30°	70 W
	60°	70 W
4Ω	–60°	127 W
	–30°	123 W
	0°	123 W
	30°	124 W
	60°	129 W
2Ω	–60°	214 W
	–30°	209 W
	0°	197 W
	30°	201 W
	60°	215 W
1Ω	–60°	309 W
	–30°	282 W
	0°	275 W
	30°	200 W
	60°	307 W



AudioGraph – The PowerCube

Amplifier : PUNCH 40I 14.4V/Mono — Rated Power: 80 W at 4 Ohms
Serial No :
Owner : ROCKFORD CORPORATION

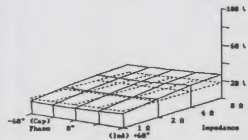
8Ω	–60°	98 W
	–30°	94 W
	0°	94 W
	30°	94 W
	60°	99 W
4Ω	–60°	184 W
	–30°	169 W
	0°	163 W
	30°	169 W
	60°	183 W
2Ω	–60°	257 W
	–30°	232 W
	0°	224 W
	30°	233 W
	60°	257 W
1Ω	–60°	243 W
	–30°	215 W
	0°	211 W
	30°	218 W
	60°	247 W

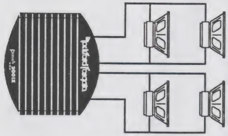


AudioGraph – The PowerCube

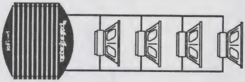
Amplifier : PUNCH 40I 12.6V Mono — Rated Power: 80 W at 4 Ohms
Serial No :
Owner : ROCKFORD CORPORATION

8Ω	–60°	90 W
	–30°	88 W
	0°	87 W
	30°	88 W
	60°	92 W
4Ω	–60°	149 W
	–30°	137 W
	0°	135 W
	30°	140 W
	60°	148 W
2Ω	–60°	205 W
	–30°	184 W
	0°	177 W
	30°	185 W
	60°	203 W
1Ω	–60°	215 W
	–30°	100 W
	0°	104 W
	30°	191 W
	60°	220 W





Punch 200ix
Method used without measurement



Punch 40i
Method used after measurement made with IM-1

Frequency Range	Punch 200ix Total Watts	Punch 40i Total Watts	Δ Watts Punch 40i	Δ dB
45-55Hz	169 Watts	173 Watts	+6 Watts	+.13dB
30-75Hz	311 Watts	241 Watts	–70 Watts	– 1.1dB

In this configuration we can achieve the same output with an Audiophile RFA-124 with either a Punch 200ix or a Punch 40i. Is the Punch 40i as powerful as the Punch 200ix? Definitely not! But, in this application, the power from the Punch 40i is being used more effectively.

If you were maximizing Punch 200ix:

$$\Delta\ dB = 10 \log \frac{755}{311} = +3.85dB \quad (30\text{-}75Hz)$$

$$\Delta\ Watts = 444\ Watts \quad (30\text{-}75Hz)$$

$$\Delta\ dB = +4.98dB \quad (45\text{-}55Hz)$$

$$\Delta\ Watts = 361\ Watts \quad (45\text{-}55Hz)$$

IM-1 Impedance Meter

The IM-1 is designed to test the impedance of speakers, passive crossover networks and speaker systems over the full audio frequency range.

Specifications

☞ Maximum Impedance Measurement:	100 Ohms
☞ Impedance Measurement Accuracy:	0 Ohms-50 Ohms: 5% 50 Ohms-100 Ohms: 10%
☞ Frequency Range:	20Hz-20kHz in 3 ranges
☞ Frequency Measurement Accuracy:	1%
☞ Typical Open Circuit Voltage At Probe Tips:	1.5 Vrms

Features

- ☞ Hand Held Operation
- ☞ Operates on 9V Battery
- ☞ Impedance Measurement Range: Full Audio Spectrum/20Hz-20kHz
- ☞ Two Operating Modes:
  - A. Frequency Dependent Impedance Measurements
  - B. Frequency Counter
- ☞ Built-In Signal Generator
- ☞ Low Battery Indicator
- ☞ Ideal for Determining Circuit Impedance and Frequency of Measurement
- ☞ Perfect for Troubleshooting and Characterizing Systems
- ☞ Protective Carrying Case

Warnings

1. NEVER connect or disconnect any component of the sound system with power applied to the system.
2. NEVER connect test leads of the IM-1 to any system that has power applied to the system.

Low Battery Indication and Battery Replacement

When the battery voltage drops too low for proper operation, the IM-1 will read "0.0" at all times in the impedance mode. The open circuit reading of the IM-1 indicates the condition of the battery during operation. The open circuit reading decreases as the battery voltage falls. When the open circuit reading decreases to about "120.0", the battery voltage becomes too low and the display reading changes to "0.0". An IM-1 with a low battery will read "0.0" in the impedance mode at all times no matter what the actual circuit impedance is. For maximum battery life, use an alkaline battery to power the IM-1.

When replacing the battery in the IM-1, use the following precautions:

1. Remove the IM-1 test leads from the circuit under test.
2. Turn off power switch on the IM-1.
3. Install battery correctly into battery connector. An incorrectly installed battery will become hot and will not allow the IM-1 to operate.

Failure to follow the battery replacement precautions may result in damage to the battery or tester.

Applications

The IM-1 measures the impedance of a circuit by supplying an A.C. current to the circuit under test and measuring the voltage developed across the circuit. Since the test signal from the IM-1 is A.C., the IM-1 can measure circuit impedance measurements, detection of open or short circuits in speaker systems, and frequency response characterization of speakers and speaker systems.

Testing Procedure

To use the IM-1, the following procedural steps are necessary to ensure accurate measurements.

1. Move the **FUNCTION** switch to the **IMPEDANCE** position.

2. Use the **FREQUENCY RANGE** switch to select the desired range for testing.
   Options are: 20Hz - 200Hz, 200Hz - 2kHz, 2kHz - 20kHz.

3. Turn the IM-1 to the ON position by rotating the **FREQUENCY ADJUSTMENT** knob clockwise until you feel a slight "click".

4. Plug the test leads into the IM-1. Be sure to observe polarity (red is +, black is –).

With the test leads open circuited (not connected to anything), the LCD display on the IM-1 will read from "120.0" to "135.0" depending on the state of charge of the internal battery.

If the display reads "0.0" check the internal battery for proper connection and state of charge. If the battery is bad replace it with a new one.

WARNING: Batteries that are in a discharged state can leak acid which can damage the IM-1.

Discharged batteries should be removed immediately. If the IM-1 is to be stored for a period of more than 30 days, the battery should be removed.

To test a speaker system/circuit use the following procedures:

1. With the IM-1 set up as described by the above procedures attach the test leads to the speaker wire that "feeds" the speaker system. The wire must be disconnected from the amplifier. It is best to check one channel at a time. i.e. Test left front then right front and so forth. Be sure to observe polarity.

2. Observe the LCD readout. A nominal impedance will be shown.
   EXAMPLE: If the speaker wire to which the IM-1 is attached feeds a speaker system/circuit that includes a 4 Ohm woofer, a 4 Ohm midrange and a 4 Ohm tweeter all wired in parallel with assorted passive crossover components, then the IM-1 readout should be approximately "3.5". This is with the FREQUENCY RANGE switch set in the 20Hz - 200Hz position and the FREQUENCY ADJUSTMENT fully counterclockwise.

3. To observe the frequency at which the impedance reading is being taken set the function switch to the FREQUENCY position.

   In the previous example, the readout should read approximately "20" when the function switch is moved to the FREQUENCY position.

4. By rotating the FREQUENCY ADJUSTMENT switch clockwise it is possible to vary the frequency within a selected range. Moving clockwise increases the frequency and counterclockwise decreases the frequency.

This allows for impedance measurements to be taken across the entire audio spectrum. With the function switch in the IMPEDANCE position, slowly rotate the FREQUENCY ADJUSTMENT knob clockwise while observing the display readout. If the impedance measurement drops below the amplifiers' "safe operating range", switch the function switch to the FREQUENCY position to isolate where the problem exists. i.e.: A low impedance measurement at 300Hz could indicate a poor crossover design, a shorted capacitor or induct coil, overlapping crossover frequencies, etc.

OSC2

Pink Noise / Sine Wave Generator

- Tuning Frequency
- System Troubleshooting
- Resonance
- Real Time Analysis (RTA) Adjustment
- Transfer Functions

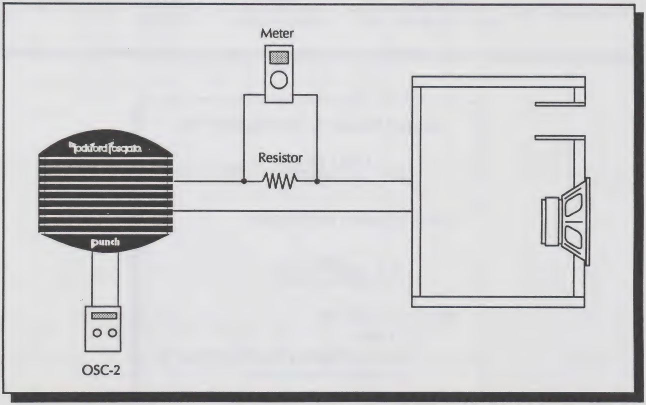
VERIFY TUNED FREQUENCY

Once you have constructed your enclosure, it is critical that you verify the actual tuned frequency of the box. Using the nomogram, or a computer program to calculate the box volume and port ratios will not guarantee exact results.

There are two methods commonly used to verify tuning. The visual method is fast and will work fine with most enclosures. This method requires an amplifier and sine wave generator.

Connect the sine wave generator to the input of the amplifier and the amplifier's output to your speaker enclosure. Set the volume so that you can see the excursion of the speaker (90-100dB). Adjust your sine wave generator frequency to 80 Hz and slowly begin to sweep down in frequency. Watching the speaker cone you will notice a null point, or the point where the speaker's excursion seems to stop. At this point, the air flow form the port will be at its maximum velocity. This is your tuned frequency.

The second method requires the addition of an AC volt meter and a resistor (50 Ohm 20 Watt). Wire the resistor in series with the speaker's voice coil and measure the AC voltage across the resistor. Adjust the amplitude until the VOM meter reads about 1-2 volts. Slowly begin to sweep down in frequency watching for a peak in the voltage. The point where voltage is at a maximum is your tuned frequency. In a vented bandpass enclosure you will have 2 peaks in the voltage, representing the two ports tuned at different frequencies.



VOLUME OF IRREGULAR SHAPE

When designing a woofer system, it is not always possible to build a regular (common) shaped enclosure. In some applications, the enclosure becomes so unique that conventional forms of volume measurement are not possible. The problem becomes: how do I measure it?

THE OLD WAY

1. Guess
   Advantage: Easy
   Disadvantage: Duh

2. Fill the enclosure full of...
   a. water
   b. popcorn
   c. sand
   Advantage: better than guessing
   Disadvantage: really messy and any material that does not take all the air space is not accurate

Example: Given: Radius = 2"
Length = 12"
Frequency of the box = 32Hz

$$\text{Volume of the box} = \frac{8466\,(2)^2}{(32)^2\,(12 + \{1.463 \times 2\})} = \frac{33864}{(1024)\,(14.926)} = \frac{33864}{15284.224} = 2.22 \text{ cubic feet}$$

Standard Formula for Vent Calculations

$$L_v = \frac{1.463 \times 10^7 \times r^2}{f_b^2\,(1728)\,V_b} - 1.463r$$

If you could stand a 0.005% error:

$$V_b = \frac{8466r^2}{f_b^2\,(L_v + 1.463r)}$$

Where: V_b in cubic feet
r^2 = radius
f_b = tuning frequency of port and enclosure
L_v = length of vent in inches

OSC2 Digital Sinewave/ Pink Noise Oscillator

The OSC2 Digital Sinewave/Pink Noise Oscillator is designed to provide a convenient means for tweaking almost every autosound installation.

Specifications

- Output Frequency: Continuously variable from 25Hz-20kHz
- Output Amplitude: Continuously variable; 700 mV RMS Max
- Distortion: 0.15%

Features

- Switchable Sinewave/Pink Noise Outputs
- Continuously Variable Sinewave Output in Three Frequency Ranges
- Adjustable Output Level
- Low Distortion
- Portable Battery Powered Operation
- Accurate Digital Readout

Warnings

To prevent possible system damage:

- NEVER connect or disconnect any component of the sound system while power is applied to the system.
- ALWAYS turn the OSC2 on BEFORE applying power to the system.
- ALWAYS turn the system off BEFORE turning the OSC2 off.
- ALWAYS verify that the volume control on the OSC2 is fully counterclockwise (minimum) BEFORE turning the system on.

Connection to the System

Connecting the OSC2 to an autosound system is as easy as plugging in 2 RCA plugs. Since the OSC2 is battery powered, there are no power wires or ground loops to contend with. The OSC2 can inject a signal anywhere line level signals are present in the system. Just unplug the existing source unit and plug in the OSC2.

Applications

Testing For Proper Phase

Set the output for sinewave and adjust the frequency between 20 and 60Hz. Slowly bring the volume up until a moderate output is obtained. Try reversing the polarity of one of the woofers. If the sound output increases, the drivers are in phase. If the output decreases, the drivers were in phase before and the polarity should be returned to its original configuration.

Adjusting Equalization

By using the OSC2's Pink Noise outputs and a Real Time Spectrum Analyzer (RTA) an autosound system can be fine-tuned for an extremely smooth response curve. Connect the OSC2 ahead of the Equalizer. Slowly bring the volume up until a moderate output is obtained. Monitor the RTA display and adjust the equalizer for the smoothest response.

Determining Resonant Frequency of the Vehicle

For people involved in Car Audio competition, the OSC2 can provide a substantial advantage. In order to obtain maximum Sound Pressure Level (SPL) scores at contests, it is important to know the resonant frequency of the vehicle. Set the OSC2 to sinewave and adjust the frequency to around 50Hz. Slowly bring the volume up until a moderate output is obtained. Now, slowly vary the frequency between 30 and 100Hz. At one point, the output of the system will increase. Observe the frequency displayed on the OSC2's digital readout. This is the resonant frequency of your vehicle. For competition purposes, select a song that contains a sustained note that very closely approximates this frequency.

Setting Crossover Points

Crossover points and midrange/tweeter polarity are easily checked with the use of the OSC2's Pink Noise outputs. As with equalization, an RTA will be necessary. Overlapping crossover points will show up as Peaks in the response curve, while underlapping crossover points will show up as Dips. Since phase shifts occur in crossover networks, it is important that midrange and tweeters that are mounted in close proximity to each other be in phase. Otherwise, Hot Spots and Holes will occur in the frequency response of the system. Try reversing the phase of either the midrange or tweeter and see if the response on the RTA smooths out at the crossover point **THIS SHOULD BE DONE PRIOR TO EQUALIZATION.**

Setting System Levels

In biamped systems, the OSC2's Pink Noise generator is useful for setting the input levels on the amplifiers. Using an RTA, adjust the input levels on the amps for the smoothest response. **THIS SHOULD BE DONE PRIOR TO EQUALIZATION.**

Troubleshooting an Installation with the OSC2

☞ Verify Operation of Source Unit

☞ Locate Ground Loops

☞ Locate Faulty RCA Cables

☞ Locate Faulty Line Level Accessories

Tuning Vented Enclosures

The OSC2 is perfect for the installer who designs and builds vented enclosures. The variable output frequency makes it simple to determine the tuning frequency of the enclosure. Slowly bring the volume up until a moderate output is obtained. Now, slowly vary the frequency from 20 to 100Hz. At the tuning frequnecy, MINIMUM cone excursion will occur.

CD #101

Low Frequency Test CD

- Setting Gain Controls
- Tuning Frequency
- Resonance
- Absolute Polarity (in a system)
- Determining F3
(low frequency system)
- Sonic Characteristics
(low frequency system)

General Information and Test Procedures

Track One: This track contains a 1kHz tone recorded at a level of –20dB relative to digital clipping. This level is normally used in digital systems to indicate "0" level on normal program material. The level of this tone provides for 20dB of headroom and is an absolute level useful for comparison with other measuring systems. This track can be used for measuring the reference loudness of a component. This is accomplished by playing Track 1 on a CD player, and then measuring the signal output voltage of the headpiece with its volume control set to maximum.

Track Two: This track begins with a 1kHz tone, recorded at –20dB relative to digital clipping. After an initial period at –20dB, the level is decreased. Eventually, the level is reduced to digital zero, which continues for the remainder of the track.

This track is excellent for observing the reproducing quality of a sound system when driven with extremely low level signals. For these tests, the volume control of the sound system should be increased to keep the perceived audio level constant. The quality of the tone should not change and become distorted or fuzzy. This is a great track in which to test the sensitive circuitry of electronic noise gates. Track 2 is a repeatable source of low level signals for initial set up and calibration of noise gates. There is no need for special equipment.

For reference, the approximate levels referenced to track time are, :01 seconds = 20dB below clipping, :20 seconds = 30dB below clipping, :27 seconds = 40dB below clipping, :35 seconds = 50dB below clipping, :47 seconds = 60dB below clipping, :51 seconds = 70dB below clipping, 1:03 seconds = 80dB below clipping, 1:08 seconds = 90dB below clipping, 1:10 seconds = limited by CD player.

CAUTION: Plenty of time has been left on the end of this track to enable one to reduce the level of the volume control. DECREASE THE VOLUME CONTROL NOW.

Track Three: Both relative, as well as absolute polarity, can be quickly and easily measured using the high-frequency pulses contained on this track. The software is compatible and designed to be used with the PD-1 Phase Detector. In order to confirm accurate readings, every fourth pulse has been inverted. Readings can only be considered valid if the checker actually indicates this reversal when making the reading.

Using the pulses on Track 3 of Test CD #101 is the ONLY accurate method of determining the absolute polarity of the entire sound system. The three positive pulses and one negative pulse begin in the D-to-A convertor of the CD player and travel through the signal cables to the intermediate components, and then down the speaker leads, through the passive crossovers and on to the speaker voice coils. Thus, the direction of motion of the speaker's cone is dependent upon the polarity of the pulse produced in the CD player.

Relative polarity must be achieved if optimum performance is expected from a sound system. A single "out-of-phase" driver can wreak disastrous consequences in an otherwise great system. All the drivers in a system must be configured so as to work together. When Track 3 is played through a sound system, and the PD-1 is positioned directly in front of every driver, identical Phase Detector responses on each driver would indicate that the system is in relative polarity.

Absolute polarity is important in a system if accurate reproduction is the goal. Absolute polarity means that the sound system is reproducing the software exactly as it was recorded. Many musical instruments and voices have non-symmetrical waveforms and must be reproduced accurately with regard to polarity. Uniform polarity within a system is also important if the harmonics are going to be reproduced uniformly with respect to their fundamentals. The three positive pulses and one negative pulse that is repeated many times on Track 3 must be detected as such for a system to be in absolute polarity. **Note:** On the PD-1, a green light indicates that the speaker is pushing towards the microphone at the head of the device. A red light indicates that the speaker is pulling away from the microphone.

Please take caution because the peak energy of this track is extremely high and could cause damage to sensitive tweeters. Keep the volume control at a modest level. The first minute of this track contains a low frequency background hum. With a normal full-range system, the level should be raised to the point where this hum is just barely audible. Take the readings very close to the individual speakers. The energy content covers the entire frequency spectrum and will give ample readings for loudspeakers covering the range from 20Hz to 20kHz.

Track Four: This track contains a slow sweep that covers the entire audible range from 20Hz to 20kHz. It is useful for making response measurements in conjunction with other test equipment. This track can also be used to check a sound system for any resonant components that may buzz or rattle during any part of the sweep. When using this track, listen carefully to the speakers for any signs of cone breakup or voice coil rubbing during the sweep.

The approximate track time vs. frequency relationship for Track 4 is as follows: :05 -=- 20Hz, :10 = 50Hz, :15 = 85Hz, :20 = 150Hz, :25 = 250Hz, :35 = 800Hz, :40 = 1.4kHz, :45 = 2.4kHz, :50 = 4.2kHz, :55 = 7.8kHz, 1:00 = 13kHz, 1:05 = 20kHz.

Track Five: Identical to Track 4 except that the sweep occurs faster. This will result in less time for the measurement.

Track Six: The sweep on this track starts at 20kHz and ends at 20Hz. It is composed of 112 discrete tones as opposed to the continuous sweeps of the previous two tracks. Track 6 is useful for specialized types of equipment designed to measure this specific signal. However, this track can also be used for some of the very same applications as Track 4 and Track 5. Note: The discrete frequencies for this disc are on file at AUTOSOUND 2000. If you require a listing, please call or write.

Track Seven: On this track there is a compilation of 16 free-running oscillators combined to generate a composite tone. The tones are totally non-synchronous in nature and are therefore more usable for spectral measurements with swept spectrum analyzers. This track can also be used to determine the overload threshold of a system particularly when equalizers are used in their boost mode. It is very easy to hear audible clipping on Track 7.

The frequencies generated in this composite are: 30Hz, 50Hz, 100Hz, 250Hz, 400Hz, 500Hz, 700Hz, 1kHz, 2.5kHz, 5kHz, 7.5kHz, 10kHz, 12.5kHz, 15kHz, 18kHz, 20kHz.

Track Eight: A 50Hz square wave tone starts this track. After a short period of time, this square wave sweeps down to 20Hz. This is useful for setting up low frequency servo systems.

Track Nine: This track contains low frequency tone bursts from 40Hz to 200Hz, in 10Hz increments. When these tones are played, they should sound sharp and clear with no hangover. Track 9 is very useful when evaluating the transient behavior of a sub-woofer system. Any tendency for the system to dull the attack of the bursts is an indication of poor transient response. Small improvements can easily be heard with this test track. The period between each burst is chosen to allow adequate time for the sound to decay away and there should be at least some total silence between bursts. Lack of this silent period is an indication of poor damping or an uncontrolled resonance in the listening area.

For a more exacting test, take a microphone and feed it directly into an oscilloscope. The actual waveform is two complete cycles of each frequency starting and stopping at zero crossover. The waveform should completely stop before starting again. Because of the dimensions of the automobile, these low frequencies are going to represent a pressure response behavior and are not likely to be bothered by standing wave phenomenon.

Tracks 10 through 98: With these tracks, the numerical track indicator of the CD player directly correlates with the frequency of the sine wave that is digitally recorded on that particular track. Track 10 is therefore 10Hz, Track 11 is 11Hz...Track 98 is 98Hz. There are no lapses between tracks and no breaks in the sine wave. Each track is approximately 30 seconds in length.

The most obvious use of these tracks would have to be determining the resonant frequency of a low frequency speaker and/or enclosure.

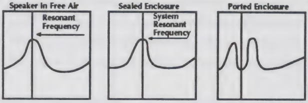
Concerning Speakers Alone

Although there are many methods of performing the measurement, the concept is that at its resonant frequency a speaker is extremely efficient. Also, the speaker will reach its maximum impedance when at resonance. Consequently, when a speaker is in resonance, the current through that speaker's voice coil will drop to its lowest level. It follows then, that the voltage across the voice coil will reach its maximum when the speaker is in resonance.

Concerning Speakers in Enclosures

A speaker in a completely sealed box has characteristics similar to a speaker in free air and will exhibit a single peak voltage across its terminals when in resonance. However, a ported box will have two peaks and a dip between those peaks. The dip between the peaks correlates to the resonant (or tuning) frequency of the ported speaker system.

It is our suggestion that an analog AC voltmeter be used to measure the voltage across the voice coil of the speaker to be tested. Also, a 50-Ohm, 20-Watt power resistor should be inserted in series with + output of the amplifier and the + terminal of the speaker.



Examples of Voltage across Speaker Terminals (Vertical) vs. Frequency (Horizontal) as displayed on Track Indicator of CD.

With the power resistor wired into place and the voltmeter connected across the terminals of the speaker, insert Test CD #101 into the CD player and set the volume loud enough to barely interfere with normal conversation. Adjust the scale on the voltmeter for a half-deflected needle.

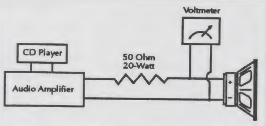
Let's say that we are to determine the resonant frequency of a 10" woofer in a completely sealed box. Start with 50Hz (Track 50) and note which way the needle deflects as the track numbers are increased through Track 51, Track 52, etc. If the voltage drops as the track numbers are increased, then the resonant frequency must be lower.

Quickly scan through Track 49, Track 48, Track 47, etc. The voltage across the speaker terminals should now be increasing indicating that you are approaching the resonant frequency. Remember that for a speaker in a completely sealed enclosure you are looking for a maximum speaker voltage on the AC voltmeter. Keep changing CD tracks until the needle stops peaking and then just begins falling back down. Some adjustment to the scale on the voltmeter may be necessary. Read the track number on the display of your CD player when the voltage is at its peak. The track number is also the resonant frequency, in Hz, of the speaker in its enclosure.

A digital multi-meter can also be used for these tests, but be aware that the low frequency response of most inexpensive DMMs leaves much to be desired. By the way, an easy way to verify the low frequency response of your DMM is to plug the signal output of the CD directly into the DMM. Using Track 4, note any variations as the signal sweeps between 20Hz to 20kHz. The AC voltage should not vary on a high quality meter. For a more detailed evaluation of your DMM, the low-frequency software on Tracks 10 through 98 can be used.

The value and wattage of the series resistor is also non-critical. The volume level of the CD player as well as the power output of the amplifier should not drastically change the results of this experiment. Also, if you prefer to measure the voltage across the series resistor (a minimum voltage would be present here at resonance), or the current flowing through the voice coil (minimum current would be present at resonance), the results should prove identical.

Track 99: Caution must be used with this track. It contains a 1kHz sine wave recorded at the highest possible level acceptable for a CD. The first and most obvious use of this track is to test the output headroom of a CD player.



Test Arrangement for Determining Resonant Frequency of a Speaker, or System, using Low Frequency CD #101.

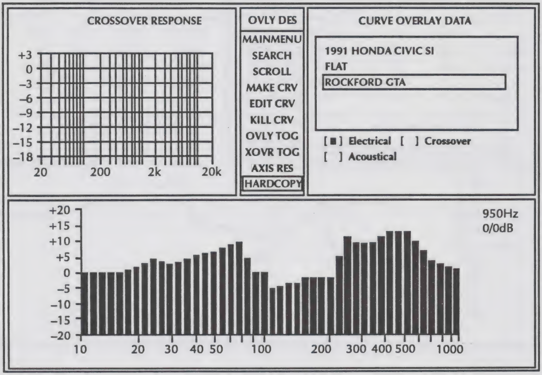
The track begins at a relatively low level and gently rises to its maximum so as to provide adequate warning. At approximately 20 seconds, the waveform actually goes into clipping and then slowly reduces in output until at :27 seconds it remains just below clipping. Track 99 remains at this level until the end of the track.

Without the need for any other test equipment, Track 99 can produce an accurate test for clipping in a CD player. Listen carefully to this track with the volume turned way down and "learn" the sound of clipping as the time indicator advances from :20 seconds to :27 seconds. Next, increase the volume until you hear a similar sound or until the volume is turned all the way up. If your unit is clipping already, it will have a tendency to mask the portion that is intentionally clipped.This procedure can be used to optimize the gain structure of an entire system.

If an oscilloscope is available, much can be discovered about the quality of the Digital-to-Analog convertor and filters within a CD player during the intentionally clipped section. A really good design will show a minimum of ringing as well as a smooth flat top on the clipped sine wave instead of very ragged uneven clipping.

TRANSFER FUNCTIONS

- Theory and Practice
- Establish a Reference
- Build a Test enclosure
- Map the Vehicle Response
- Import the Data into Term-Pro


CROSSOVER RESPONSE
+3
0
-3
-6
-9
-12
-15
-18
20
200
2k
20k
OVLY DES
MAINMENU
SEARCH
SCROLL
MAKE CRV
EDIT CRV
KILL CRV
OVLY TOG
XOVR TOG
AXIS RES
HARDCOPY
CURVE OVERLAY DATA
1991 HONDA CIVIC SI
FLAT
ROCKFORD GTA
[■] Electrical [ ] Crossover
[ ] Acoustical
+20
+15
+10
+5
0
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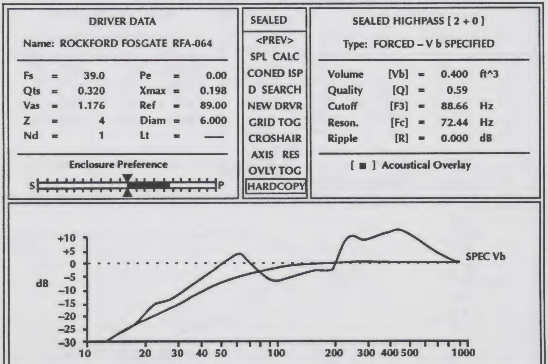
10
20
30
40
50
100
200
300
400
500
1000
950Hz
0/0dB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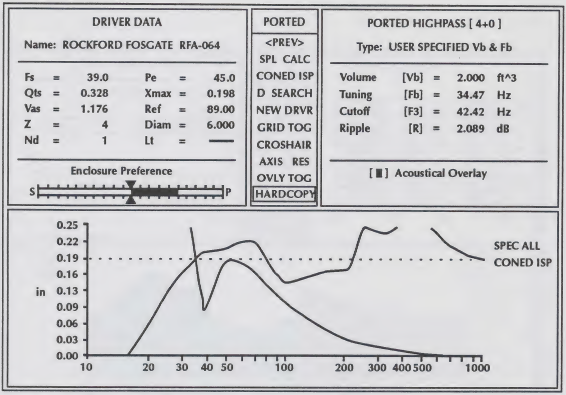
Pe = 0.00
Qts = 0.320
Xmax = 0.198
Vas = 1.176
Ref = 89.00
Z = 4
Diam = 6.000
Nd = 1
Lt = —
Enclosure Preference
S
P
SEALED
<PREV>
SPL CALC
CONED ISP
D SEARCH
NEW DRVR
GRID TOG
CROSHAIR
AXIS RES
OVLY TOG
HARDCOPY
SEALED HIGHPASS [2 + 0]
Type: FORCED – V b SPECIFIED
Volume [Vb] = 0.400 ft^3
Quality [Q] = 0.59
Cutoff [F3] = 88.66 Hz
Reson. [Fc] = 72.44 Hz
Ripple [R] = 0.000 dB
[■] Acoustical Overlay
+10
+5
0
-5
-10
-15
-20
-25
-30
dB
10
20
30
40
50
100
200
300
400
500
1000
SPEC Vb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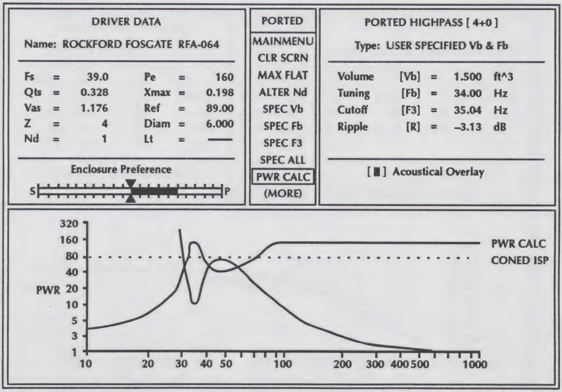
Xmax = 0.198
Ref = 89.00
Diam = 6.000
Lt = ——
Enclosure Preference
S
P
PORTED
<PREV>
SPL CALC
CONED ISP
D SEARCH
NEW DRVR
GRID TOG
CROSHAIR
AXIS RES
OVLY TOG
HARDCOPY
PORTED HIGHPASS [4+0]
Type: USER SPECIFIED Vb & Fb
Volume [Vb] = 2.000 ft^3
Tuning [Fb] = 34.47 Hz
Cutoff [F3] = 42.42 Hz
Ripple [R] = 2.089 dB
[■] Acoustical Overlay
0.25
0.22
0.19
0.16
in 0.13
0.09
0.06
0.03
0.00
10 20 30 40 50 100 200 300 400 500 1000
SPEC ALL
CONED ISP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Xmax = 0.198
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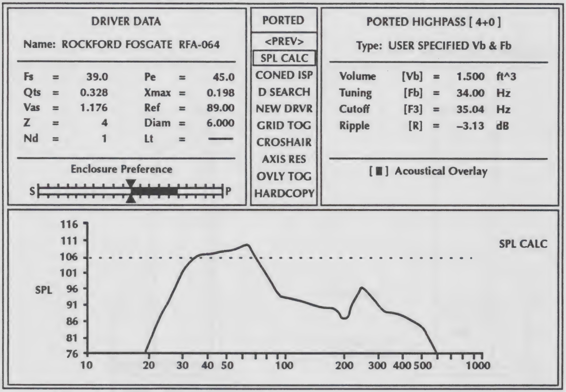
Diam = 6.000
Lt = ——
Enclosure Preference
S
P
PORTED
MAINMENU
CLR SCRN
MAX FLAT
ALTER Nd
SPEC Vb
SPEC Fb
SPEC F3
SPEC ALL
PWR CALC
(MORE)
PORTED HIGHPASS [4+0]
Type: USER SPECIFIED Vb & Fb
Volume [Vb] = 1.500 ft^3
Tuning [Fb] = 34.00 Hz
Cutoff [F3] = 35.04 Hz
Ripple [R] = –3.13 dB
[■] Acoustical Overlay
320
160
80
40
PWR 20
10
5
3
1
10 20 30 40 50 100 200 300 400 500 1000
PWR CALC
CONED ISP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Vas = 1.176
Z = 4
Nd = 1
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Xmax = 0.198
Ref = 89.00
Diam = 6.000
Lt = ——
Enclosure Preference
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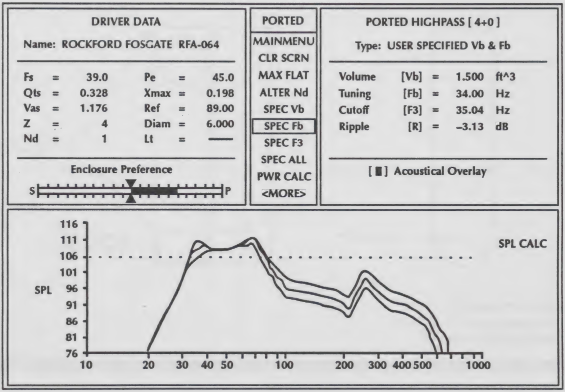
SPL CALC
CONED ISP
D SEARCH
NEW DRVR
GRID TOG
CROSHAIR
AXIS RES
OVLY TOG
HARDCOPY
PORTED HIGHPASS [4+0]
Type: USER SPECIFIED Vb & Fb
Volume [Vb] = 1.500 ft^3
Tuning [Fb] = 34.00 Hz
Cutoff [F3] = 35.04 Hz
Ripple [R] = -3.13 dB
[■] Acoustical Overlay
SPL CALC
SPL
116
111
106
101
96
91
86
81
76
10
20
30
40
50
100
200
300
400
500
1000


DRIVER DATA
Name: ROCKFORD FOSGATE RFA-064
Fs = 39.0
Qts = 0.328
Vas = 1.176
Z = 4
Nd = 1
Pe = 45.0
Xmax = 0.198
Ref = 89.00
Diam = 6.000
Lt = ——
Enclosure Preference
S
P
PORTED
MAINMENU
CLR SCRN
MAX FLAT
ALTER Nd
SPEC Vb
SPEC Fb
SPEC F3
SPEC ALL
PWR CALC
<MORE>
PORTED HIGHPASS [4+0]
Type: USER SPECIFIED Vb & Fb
Volume [Vb] = 1.500 ft^3
Tuning [Fb] = 34.00 Hz
Cutoff [F3] = 35.04 Hz
Ripple [R] = -3.13 dB
[■] Acoustical Overlay
SPL CALC
SPL
116
111
106
101
96
91
86
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76
10
20
30
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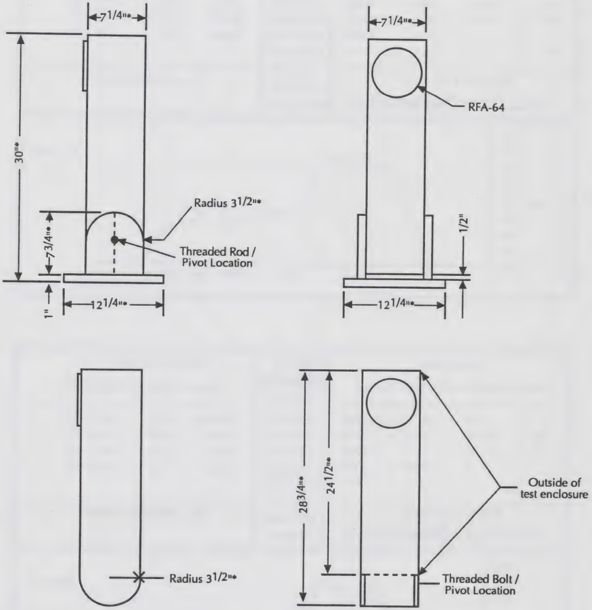
300
400
500
1000

Test Enclosure RFA-64



*Includes 1/8" Carpet thickness.
Total Volume approx. 0.48 cu. ft.
Enclosure wall thickness 1/2"

Transfer Function Data Worksheet

Vehicle: ______________________________ Date: ____________

Notes: ____

____

____

____

Frequency in Hertz	Applied AC Voltage (Measured	Measured SPL (In Car)	Converted Power (Calculated)	Calculated SPL (Term-Pro)	Transfer Function (Δ SPL)
11Hz					
12Hz					
13Hz					
14Hz					
16Hz					
18Hz					
19Hz					
22Hz					
24Hz					
26Hz					
29Hz					
32Hz					
36Hz					
40Hz					
44Hz					
49Hz					
54Hz					
60Hz					
66Hz					
74Hz					
81Hz					
90Hz					
98Hz					

Transfer Function
Data Worksheet Guideline

Step 1: Frequency in Hertz (Measured)

This is the test frequency output from your source unit. Frequencies from 10Hz to 98Hz are available from the Autosound 2000 Test CD #101. For frequencies above 98Hz, use the OSC2.

Step 2: Applied AC Voltage (Measured)

This is the AC voltage output from your amplifier. Use a digital volt meter and measure the voltage output of your amplifier at the test enclosure. Keep in mind not to exceed the maximum excursion capable of the test speaker. For best results, set the output level starting form the lowest frequency testing point, taking care not to exceed the maximum excursion capable of the test speaker. Select the desired frequency testing point while leaving the output of a fixed level.

Step 3: Measured SPL (In Car)

This is the measured SPL in the vehicle at the test location. This measurement is at the test frequency with the applied AC Voltage selected during testing.

Step 4: Converted Power (Calculated)

This is a conversion formula for data input into Term-Pro. Actually, Term-Pro calculates the SPL of a speaker/enclosure based off the AC Voltage applied to the speaker. Term-Pro relates this information (accepts and displays the information) as input power and/or power applied. To convert from AC Voltage over to Power (Watts), take the square of the input voltage and divide that by the nominal impedance rating of the speaker. This is the applied power used to determine the Calculated SPL in Term-Pro.

For a 4Ω Driver

$$P = \frac{(\text{Measured AC Voltage})^2}{4}$$

For an 8Ω Driver

$$P = \frac{(\text{Measured AC Voltage})^2}{8}$$

Step 5: Calculated SPL (Term-Pro)

This is the SPL predicted by Term-Pro in a free-field environment. The accuracy of this measurement will depend on the accuracy of the sensitivity rating by the manufacturer. Input the Converted Power (in Watts) from the Converted Power section into the SPL calculation of Term-Pro. Be sure the enclosure you use accurately tracks (resembles) the response of your test enclosure. For more information on selecting the Term-Pro test enclosure see the section "Build Your Reference Test Enclosure."

Step 6: Transfer Function (ΔSPL)

To determine the Transfer Function, subtract the Calculated SPL (ΔSPL) from the Measured SPL (in car). The difference between the two is the Transfer Function (or Acoustical Gain) of the vehicle. This information can be imported into the Overlay Design Section of Term-Pro for use when designing enclosures. If you do not have Term-Pro or another similar enclosure design program (shame on you). This information can be used as a reference material when choosing a box design of a speaker location.

Transfer Function Data Worksheet

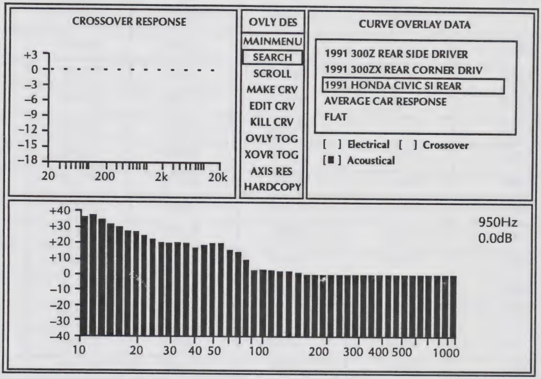
Vehicle: '91 Civic S1 Rear

Date: 11/12/93

Notes: ____________________

Frequency in Hertz	Applied AC Voltage (Measured	Measured SPL (In Car)	Converted Power (Calculated)	Calculated SPL (Term-Pro)	Transfer Function (Δ SPL)
11Hz	3.53V	104dB	3.16W	69.5dB	+34.5dB
12Hz	3.60V	106dB	3.24W	71.1dB	+34.9dB
13Hz	3.68V	106dB	3.39W	72.4dB	+33.6dB
14Hz	3.74V	106dB	3.50W	73.6dB	+32.4dB
16Hz	3.79V	106dB	3.59W	75.3dB	+30.7dB
18Hz	3.83V	105dB	3.67W	77.4dB	+27.6dB
19Hz	3.79V	105dB	3.59W	80.3dB	+24.7dB
22Hz	3.79V	105dB	3.59W	82.0dB	+22.0dB
24Hz	3.62V	104dB	3.28W	82.8dB	+21.2dB
26Hz	3.82V	104dB	3.65W	84.5dB	+20.5dB
29Hz	3.80V	105dB	3.61W	84.5dB	+205dB
32Hz	3.79V	107dB	3.59W	85.3dB	+21.7dB
36Hz	3.72V	107dB	3.46W	87.0dB	+20.0dB
40Hz	3.70V	105dB	3.42W	87.8dB	+17.2dB
44Hz	3.63V	108dB	3.29W	88.6dB	+19.4dB
49Hz	3.60V	110dB	3.24W	89.5dB	+20.5dB
54Hz	3.55V	110dB	3.15W	90.3dB	+19.7dB
60Hz	3.48V	106dB	3.03W	91.1dB	+14.9dB
66Hz	3.45V	104dB	2.98W	92.0dB	+12.0dB
74Hz	3.27V	101dB	2.67W	92.8dB	+ 8.2dB
81Hz	3.36V	96dB	2.82W	92.8dB	+ 3.2dB
90Hz	3.34V	97dB	2.79W	93.6dB	+ 3.4dB
98Hz	3.30V	98dB	2.72W	94.5dB	+ 3.5dB

Overlay curve from test data.



SYSTEM LEVEL MATCHING

- Reasons for Level Matching
- Dynamic Range of a System
- Using a Reference
- Level Setting Process

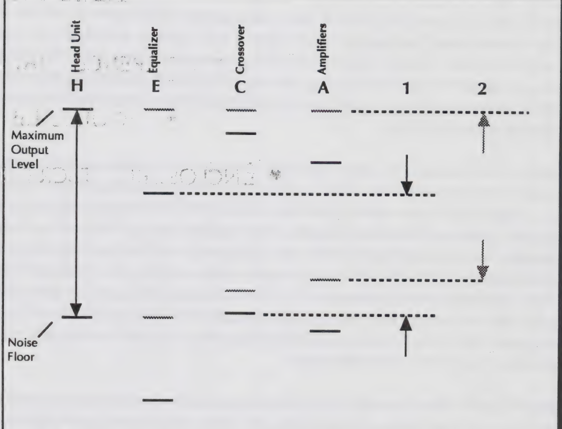
SETTING SYSTEM GAINS

Once the system is up and running, the first step to maximizing the sonic potential and minimizing the possibility of system failure is to properly adjust the System Gain Structure. By adjusting the system gains, one is able to maximize the undistorted output while minimizing the noise floor. This relationship is referred to as the signal to noise ratio or the dynamic range of the system. Many people believe that system gains do not need to be set if you are using components from only one manufacturer. More than 99 times out of 100 this will not be the case.

In an audio system, the dynamic range of a system is determined by the maximum undistorted output of the system relative to the noise floor. This noise floor can be interpreted in two different ways. The floor could be considered to be the ambient noise floor of the vehicle. This value is measured in dB and is determined by measuring the ambient noise in the vehicle with the stereo off. (This value for the noise floor will obviously increase when the vehicle is driven on the highway, etc.) Another noise floor, system noise floor, is also measured in dB and is determined by measuring the ambient noise of the sound system with the system's volume adjusted to the lowest position. In the IASCA and USAC forums, the ambient noise floor is determined using a recorded disk track of 0-bits, the technical term for no sonic information recorded, with the system volume control adjusted to the highest position. Any sound emanating from the sound system is called "system hiss" and is undesirable.

If the gain structure is adjusted properly, the dynamic range of the entire system is determined by the signal-to-noise ratio of the worst component in the audio chain. If the system's gains are not adjusted properly, the dynamic range can be much worse to nonexistent. Figure 1 illustrates a normal system set with factory settings. The top bar references the maximum output of the component and the bottom bar is the noise floor of each of the components. The space in between is the dynamic range of the component. The dynamic range of the system is determined by the lowest output and the highest noise floor.

The object of setting gains is to adjust all the audio components' input and output gain structure (whichever they have) so each component can pass the maximum signal that is undistorted. By doing this, each component's capabilities will be matched with the components that precede and follow it in the audio chain.



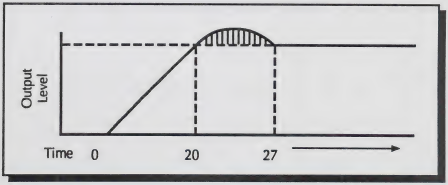
1 Dynamic Range of an unadjusted system (system left at factory gain settings)

2 Dynamic Range of system with gain levels

Figure 1

OUTPUT FROM CD101 (TRACK 99)

(0dB is the maximum undistorted output available from CD recordings)



From 20-27 seconds, track 99 is digitally clipped to give users a refrence to what clipping looks/sounds like.

APPENDIX

- Reference Material
- Source Listing
- Enclosure Descriptions

Advanced RTTI Source List

Below is a list of equipment needed to perform the tasks explained in the Advanced RTTI Training Seminar.

Rockford Equipment

- PD1 Phase Detector
- PD2 Positive Pulse Generator
- IM-1 Impedance meter
- OSC2 Sine Wave/Pink Noise Generator
- CD #101 Test Disc
- PD-TAPE Positive/Negative Pulse Recorded Tape
- TERM-PRO Computer Aided Software Program

Radio Shack Equipment

- #277-1008 Archer Mini Audio Amplified Speaker
- #271-045 100kΩ 1/2 watt carbon resistor
- #271-1347 100kΩ 1/4 watt resistor 1%
- #274-378 3.5mm to RCA Adaptor
- #274-395 3.5mm to RCA Adaptor 90° Right Angle
- #271-309 Asst. 1/4 Watt Metal Film Resistors

Other Products

- Digital Volt/Ohm Meter
- Pocket Calculator having the following functions
  - Multiplication
  - Division
  - Addition
  - Subtraction
  - Exponents (Square Roots, etc.)
  - Logarithms
  - Scientific Notation
- Source for resistors 5% or better (1% 1/4 watt preferred)

Term-Pro Requirements

Term-Pro is designed for use with IBM PC/XT/AT or compatible computers.
Below are the system requirements for the software.

Computer:	IBM PC/XT/AT or 100% compatibles
Driver:	Hard disk
Memory:	640k RAM
Graphics:	CGA, EGA, or VTA (EGA or VGA required for color)
System:	MS-DOS 3.0 or later
Printer:	Optional - Epson or Epson compatible dot matrix HP Laserjet II or compatible
Mouse:	Optional - Microsoft or Microsoft compatible

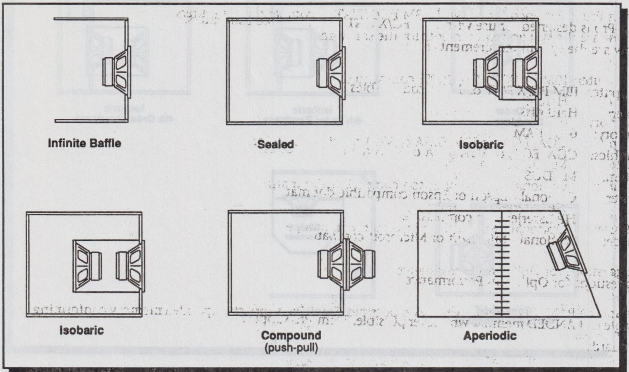
Suggestions for Optimum Performance

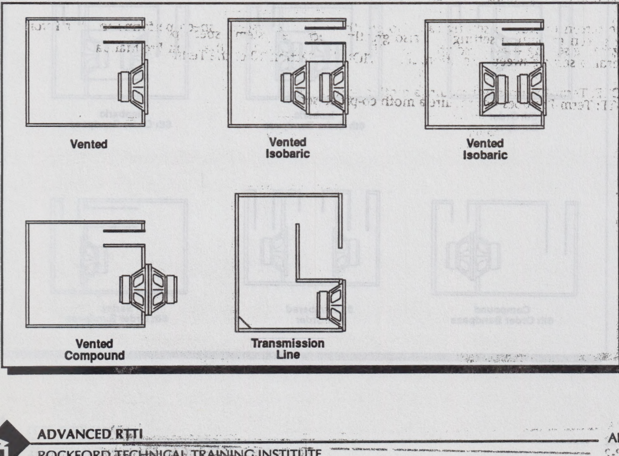
Enable EXPANDED memory whenever possible. Term-Pro supports expanded memory conforming to LIM standards.

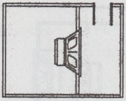
Program performance will be enhanced when using an EGA or VGA graphics adapter. In addition to high-resolution color graphics, program execution speed will increase substantially as well.

The screen resolution setting can also greatly affect the system's speed performance. For information regarding setting sweep resolution, see OPTIONS in Section 10 of the Term-Pro Manual.

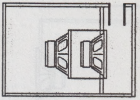
NOTE: Term-Pro does not require a moth co-processor.


Infinite Baffle
Sealed
Isobaric
Isobaric
Compound
(push-pull)
Aperiodic

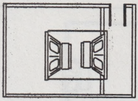

Vented
Vented
Isobaric
Vented
Isobaric
Vented
Compound
Transmission
Line



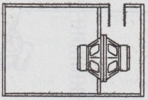
4th Order
Bandpass



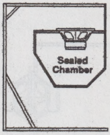
Isobaric
4th Order Bandpass



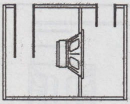
Isobaric
4th Order Bandpass



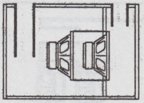
Compound
4th Order Bandpass



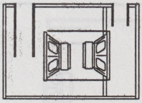
Folded Horn



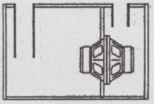
6th Order
Bandpass



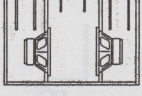
Isobaric
6th Order Bandpass



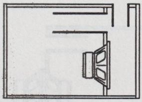
Isobaric
6th Order Bandpass



Compound
6th Order Bandpass



3 Chambered
6th Order



Series
6th Order Bandpass